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INVESTIGATION OF CONTINUOUSLY VARIABLE SLOPE DELTA MODULATOR/DE--ETC(U)
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INVESTIGATION OF CONTINUOUSLY VARIABLE SLOPE
DELTA MODULATOR/DEMODULATOR COMPATABILITY

THESIS

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Capt USAF

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INVESTIGATION OF CONTINUOUSLY VARIABLE SLOPE DELTA MODULATOR/DEMODULATOR COMPATABILITY

THESIS

Presented to the Faculty of the School of Engineering
of the Air Force Institute of Technology
Air University
in Partial Fulfillment of the
Requirements for the Degree of
Master of Science

by

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Graduate Electrical Engineering

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Preface

In this thesis I have sought to create a computer model of the NATO standard continuously variable slope delta voice encoding system with sufficient flexability to permit continued study of the standard's specifications and tolerances. This investigation has started the process of evaluating the proposed NATO standard, however, additional study is necessary to determine the standard's adequacy to assure system interoperability.

I wish to thank my thesis advisor, Capt. Kizer, and the members of the thesis committee, Lt. Col. Carpinella and Capt Seward, for their assistance, guidance, and tolerance during the course of this project.

Jeffrey A. Lersch

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Abstract

A computer model of the continuously variable slope delta voice encoding system specified in the draft STANAG on "Analogue/Digital Conversion of Speech Signals for Tactical, Digital, Area Communications Systems", dated June 1978, is developed and implemented in FORTRAN IV. The model's performance is then characterized in terms of idle channel noise, total harmonic distortion, intermodulation distortion, signal-to-noise ratio, and frequency response. For each of these attributes, the system's performance is presented graphically and compared to the criteria established in the draft standard. The model is then exercised by varying the system parameters to the limits imposed by the standard and the resulting performance compared to the previously determined ideal system performance. The results show that the performance characteristics measured are most sensitive to the primary integrator response and output filter response when the system parameters are restricted to the range allowed by the draft NATO standard.

INVESTIGATION OF CONTINUOUSLY VARIABLE SLOPE DELTA (CVSD) MODULATOR/DEMODULATOR COMPATABILITY

I. Introduction

A draft NATO standard on the analog to digital conversion of speech signals using continuously variable slope delta (CVSD) modulation is presently being circulated among the military services for comments. The proposed standard, "The Analogue/Digital Conversion of Speech Signals for Tactical, Digital, Area Communications Systems," June 1978, seeks to assure transmission systems interoperability by standardizing the system architecture and setting tolerances on key system parameters. The draft standard (see Appendix A) consists mainly of end-to-end system performance criteria, primarily signal-to-noise ratios and amplitude response characteristics. No standards are specifically established for transmission-end/reception-end mismatch performance.

The Air Force Communications Command, AFCC/OA, has questioned whether the limited number of specifications given are adequate to assure system performance when the CVSD encoding equipment is not perfectly matched to the decoding equipment. Are the tolerances specified sufficiently narrow to assure no serious signal degradation when the modulator and demodulator parameters differ by the maximum amount allowed by the draft standard? This is the question that this investigation seeks to answer.

Problem Statement Determine the adverse effects on the transmitted signal and their severity when the CVSD encoder and decoder parameters differ within the limits allowed by the draft STANAG on "The Analogue/Digital Conversion of Speech Signals for Tactical, Digital, Area Communications Systems," June 1978.

Approach The approach of this investigation is to perform a computer simulation of the CVSD analog to digital conversion system then evaluate the system's performance under varying external and internal conditions. Initially, a basic mathematical analysis of the system components is performed and mathematical models of the CVSD encoder, decoder and the

input and output filters are developed. These models are then translated into computer subroutines and coded in FORTRAN. In the next section, the tests used to characterize the model are described. These tests consist of the standard voice frequency measurements as, idle channel noise, total harmonic distortion, intermodulation distortion, signal-to-noise ratio, and frequency response. The system is first characterized with the encoder and decoder parameters matched. Each test is performed at frequencies and amplitudes across the normal active range of the system. After system performance under ideal conditions is established, the system parameters are allowed to vary across the ranges allowed by the draft standard and the degradation of the transmitted signal by encoder and decoder parameter mismatching evaluated. The results of the testing are then analyzed to determine which parameter mismatches most seriously degrade system performance and to determine if the degradation is serious enough to prevent signal transmission.

II. Analog/Digital Conversion System Model

The basic analog/digital CVSD system defined in the draft standard is shown in the block diagram in Figure 1. It consists of four major components, the input and output filters, the CVSD encoder, and the CVSD decoder. Each of these components is discussed in the following sections.

CVSD Encoder Operation The CVSD encoder structure is shown in Figure 2. The bandlimited signal from the input low-pass filter is applied to one input of the comparator and sampled at the clock rate, either 16 or 32 kb/s. Each input sample is compared to an estimate of the signal generated by the encoder feedback network from previous input samples. In this model, the comparator output is +1 if the input sample is greater than the signal estimate and -1 if the input sample is less than the estimate. This polar signal is converted to binary (+1 = 1, -1 = 0) and forms the transmitted data signal. To generate the next signal estimate, the polar signal from the comparator is routed to the input of the slope overload detector.

Slope overload, as defined for this system, is the condition when the last three comparator outputs are identical, either all +1's or all -1's. This indicates that the input signal amplitude is rising or falling, respectively, faster than the encoder can track using the present step size. Other systems define slope overload by different length strings of identical comparator outputs. Strings of two or four identical bits are also commonly used to indicate slope overload. The last three comparator outputs are stored in a shift register within the slope overload detector and combinational logic circuits used to determine if a slope overload condition exists. The slope overload detector output controls the position of the switch shown in the block diagram. Under normal conditions, when slope overload does not exist, the switch is in the V_{min} position. Upon occurrence of slope overload, the switch position is changed and V_{max} is applied to the input of the syllabic filter.

The syllabic filter is generally a simple single pole RC filter

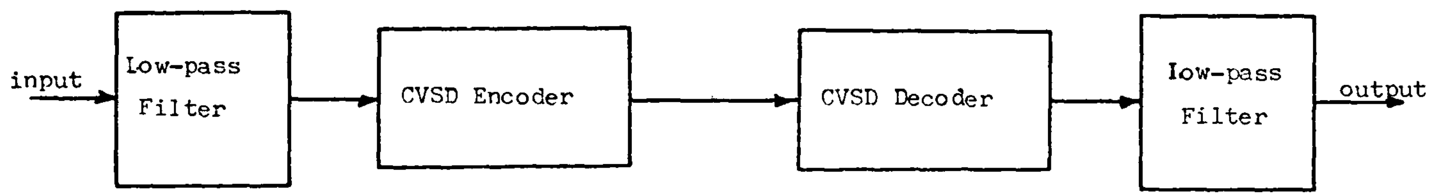


Figure 1. The CVSD Signal Processing System

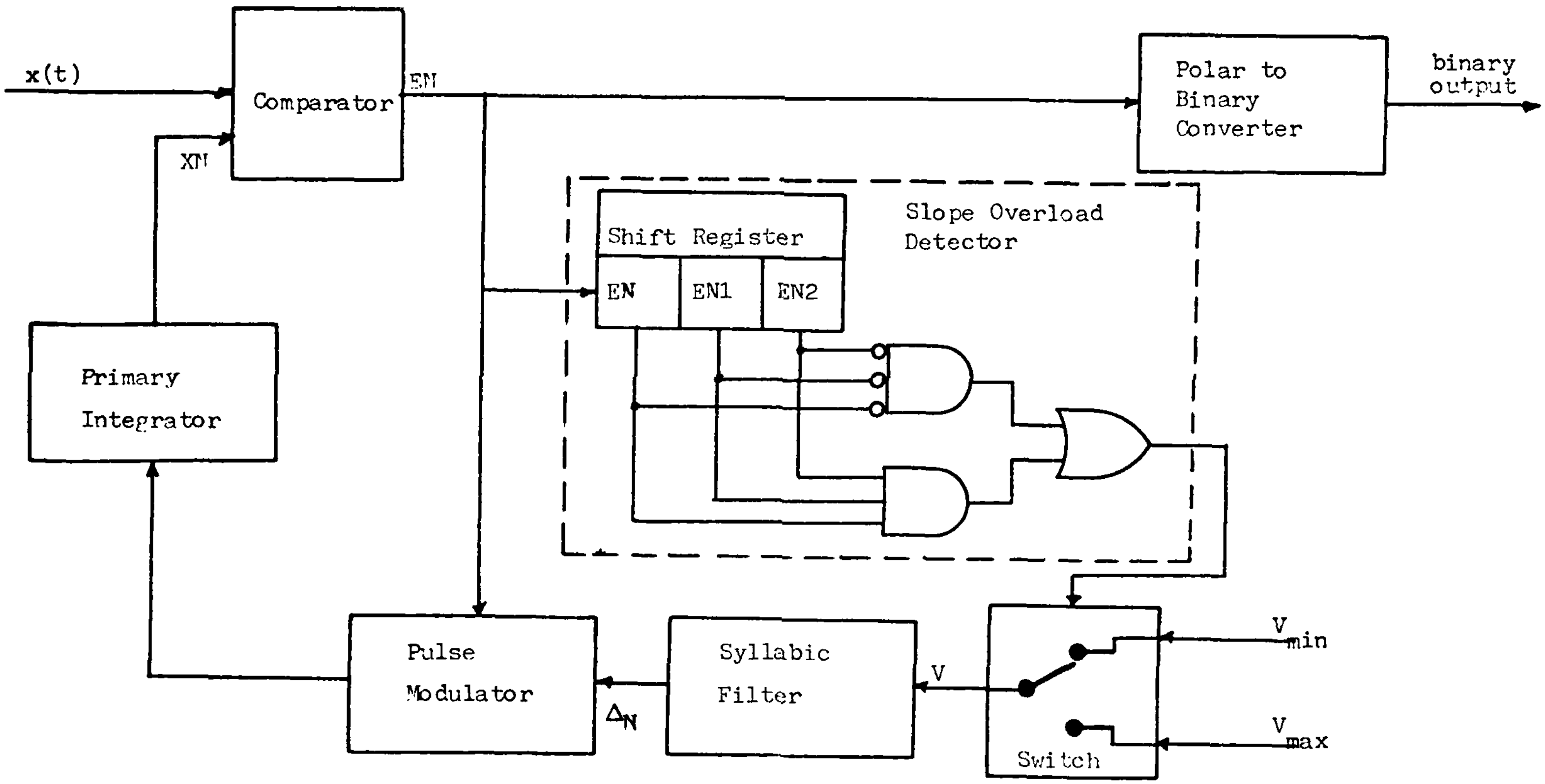


Figure 2. CVSD Encoder Block Diagram

whose output is defined as the step size of the CVSD encoder. The syllabic filter controls the response of the system to the envelope of the speech signal being processed. Prolonged application of V_{min} to the input of the syllabic filter causes the output to decrease to a minimum non-zero value that approaches V_{min}. Under continuous slope overload conditions, V_{max} is continuously applied to the input of the syllabic filter causing the filter output to increase to an average value approaching V_{max}. The magnitude of the syllabic filter output is used to control the amplitude of the output pulse of the pulse modulator. Polarity of the pulse is controlled by the last output of the comparator.

The primary integrator responds to the square wave signal from the pulse modulator and its output forms the signal estimate used by the comparator. At the end of each clock period a new estimate is available to be used by the comparator in generating the next binary output and the next signal estimate. The primary integrator's response controls the maximum analog signal frequency that can be processed through the the CVSD analog to digital conversion system. In figure 3, are shown sample waveforms at each stage of the analog to digital conversion process.

Encoder Algorithm The mathematical description of the CVSD encoder operation is largely a description of its component filters, the primary integrator and the syllabic filter. One of the system characteristics specified by the draft standard is the primary integrator response. The impulse response, in its simplist form, is given as,

$$\alpha(t) = e^{-2\pi f_{c1} t} \qquad (1)$$

where

f_{c1} = the pole frequency of the filter in hertz

The primary integrator input signal is the square wave output of the pulse modulator, which for a single pulse can be described as,

$$a(t) = 0 \qquad t < 0 \text{ and } t > T$$
$$= a \qquad 0 \leq t \leq T$$

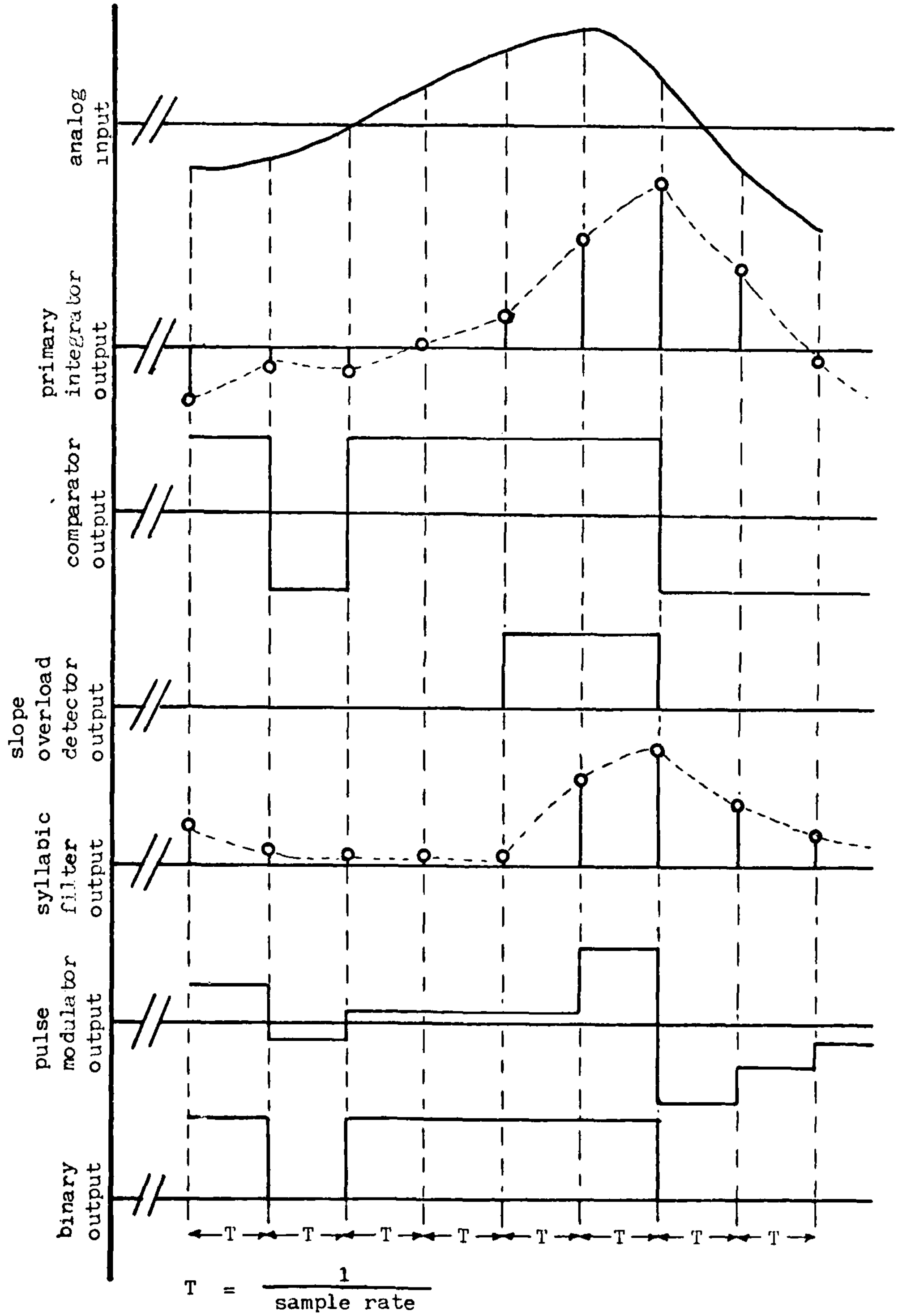


Figure 3. Sample Waveforms at Various Points in the Encoder

where

$$T = \frac{1}{\text{sample rate}}$$

The primary integrator output is determined by convolving the filter impulse response with the input signal.

$$x(t) = a(t) * \alpha(t) = \int_{-\infty}^{\infty} a(\tau)\, \alpha(t - \tau)\, d\tau$$

$$= 0\,, \quad \text{for } t < 0$$

$$= \frac{a}{2\pi f_{c1}} \left[1 - e^{-2\pi f_{c1} t}\right], \quad \text{for } 0 \le t \le T \qquad (3)$$

$$= \frac{a}{2\pi f_{c1}} \left[1 - e^{-2\pi f_{c1} T}\right] e^{-2\pi f_{c1}(t - T)}\,, \quad \text{for } t > T$$

Since the primary integrator output is of interest only at the end of each clock period, when it is used for comparison with the input analog signal, the continuous equations developed above can be simplified as follows. For $t = nT$,

$$x_n = \frac{a}{k_1} (1 - \alpha)\, \alpha^{n-1}\,, \quad n = 1,2, \ldots, N \qquad (4)$$

where

$$\alpha = e^{-2\pi f_{c1} T}$$

$$k_1 = 2\pi f_{c1}$$

Using superposition, the primary integrator output as the result of a series of N pulses can be described as,

$$x_N = \frac{1}{k_1} \left[a_N (1 - \alpha) + a_{N-1} (1 - \alpha)\alpha + \cdots + a_1 (1 - \alpha)\alpha^{N-1} \right]$$

$$= \sum_{n=0}^{N-1} \frac{a_{N-n}}{k_1} (1 - \alpha)\, \alpha^n\,, \quad \text{for } n = 1,2, \ldots, N \qquad (5)$$

This expression can also be defined recursively, depending only on the present input and the last output. This definition can be used to simulate the CVSD encoder on a computer.

$$x_N = x_{N-1}\alpha + (1 - \alpha)\frac{a_N}{k_1} \tag{6}$$

The analysis of the syllabic filter output follows identically that of the primary integrator. The impulse response of the syllabic filter is,

$$\beta(t) = e^{-\left[\frac{2\pi t}{t_c}\right]} \tag{7}$$

where

t_c = the time constant of the syllabic filter

The recursive expression for the syllabic filter output is,

$$\Delta_N = \Delta_{N-1}\beta + (1 - \beta)\frac{V_N}{k_2} \tag{8}$$

where

$k_2 = \frac{2\pi}{t_c}$

$\beta = \exp\left[-\frac{2\pi T}{t_c}\right]$

V_N = either V_{max} or V_{min}, the syllabic filter input

Encoder Computer Subroutine   Equations (6) and (8) are implemented in the subroutine used to perform the CVSD encoding for this investigation. Figure 3 is a flowchart of the subroutine used for encoding and Appendix B is the FORTRAN code used. All of the system defining parameters are transmitted to the subroutine through the calling statement. Encoding is performed on an array basis. The analog signal to be analog to digital converted is first sampled at the clock rate and the samples placed in the input array, which is of size 1 x N, where N is the number of samples. All of the samples are encoded by the subroutine and the binary data stream placed in the output array before the subroutine returns control

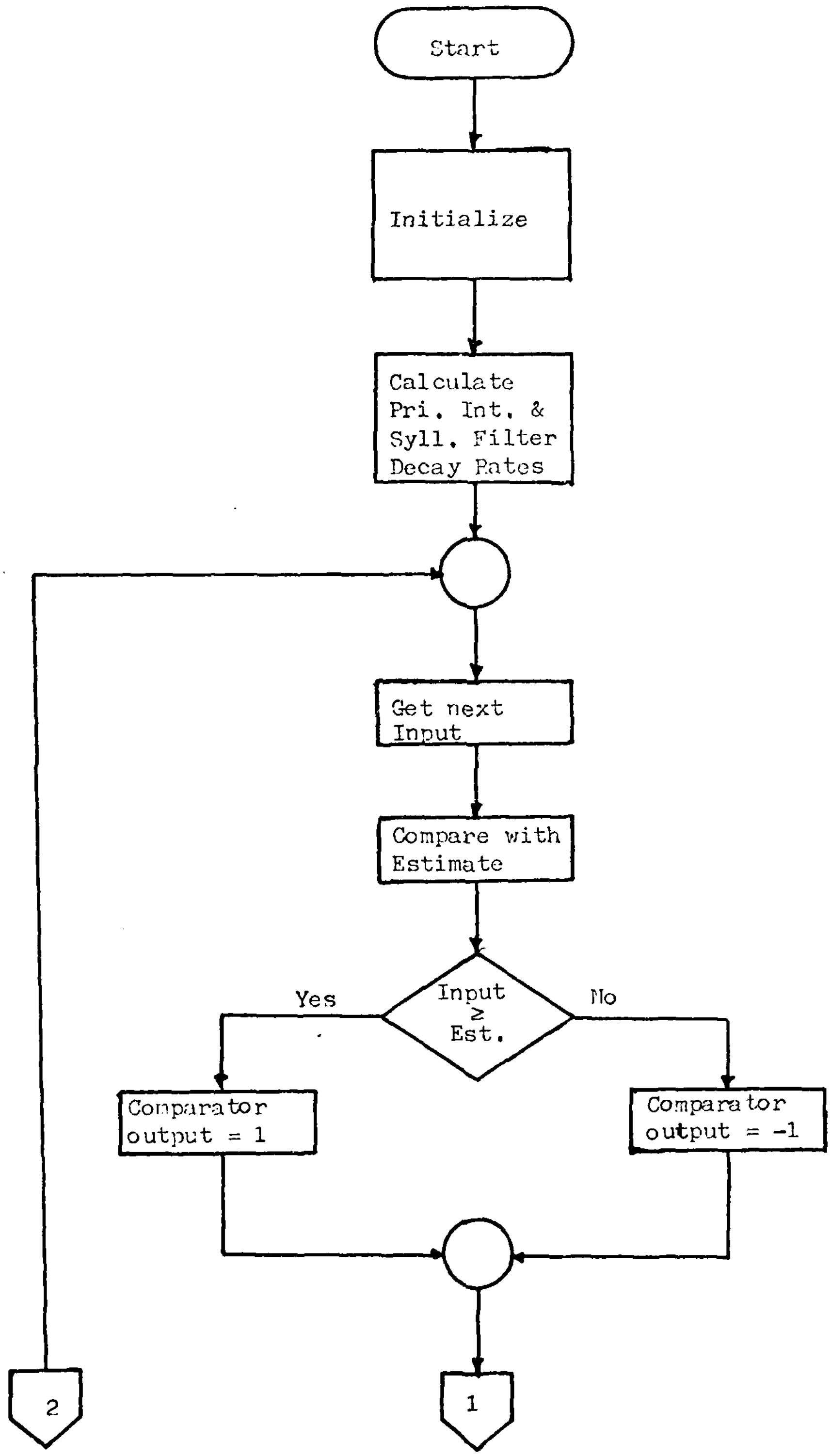


Figure 4. CVSD Encoder Subroutine Flowchart

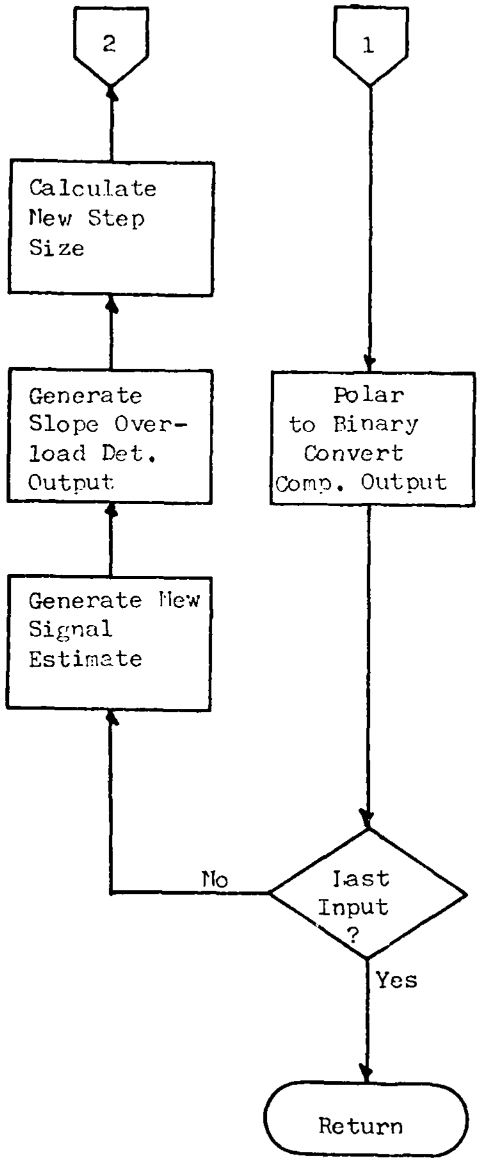


Figure 4 (continued). CVSD Encoder Subroutine Flowchart

to the calling program.

Two of the primary system defining parameters, VMAX and VMIN must be generated by subroutine VMAXOPT (appendix I) before the encoder subroutine is called. It should be noted that the constants k_1 and k_2 derived in equations (5) and (8) are not specifically included in the program statements defining the primary integrator and syllabic filter responses but are expected to be included in the values calculated for VMAX and VMIN.

The variable DC in the subroutine is the duty cycle of the slope overload detector. This variable is not used during the encoding process. Instead, it is used only by VMAXOPT when the values of VMAX and VMIN are being determined.

CVSD Decoder Operation The CVSD decoder circuit is identical to the encoder feedback circuit. A block diagram of the decoder is shown in figure 5. The only difference between the decoder and the encoder is that the decoder has no comparator. The binary signal from the encoder is applied directly to the slope overload detector and the output signal is taken from the primary integrator. The signal estimate generated in the decoder is identical to that generated in the encoder, if the parameters of each unit are identical. However, at the decoder the signal estimate is of interest at all times and not just at the sample periods, as the decoder signal estimate is the approximation of the analog signal transmitted by the CVSD encoder. Figure 6 shows sample waveforms at various points within the decoder. The waveforms are identical to those shown in figure 3, except that the decoder primary integrator output is shown as a continuous signal.

Decoder Algorithm Since the decoder circuit is identical to the encoder circuit without the comparator, the mathematical analysis developed for the encoder is also applicable to the decoder. One exception, however, is that the simplification used to obtain equation (4) is not generally applicable to the decoder since the primary integrator output of the decoder is required to be continuous. The recursive expression for the decoder output is,

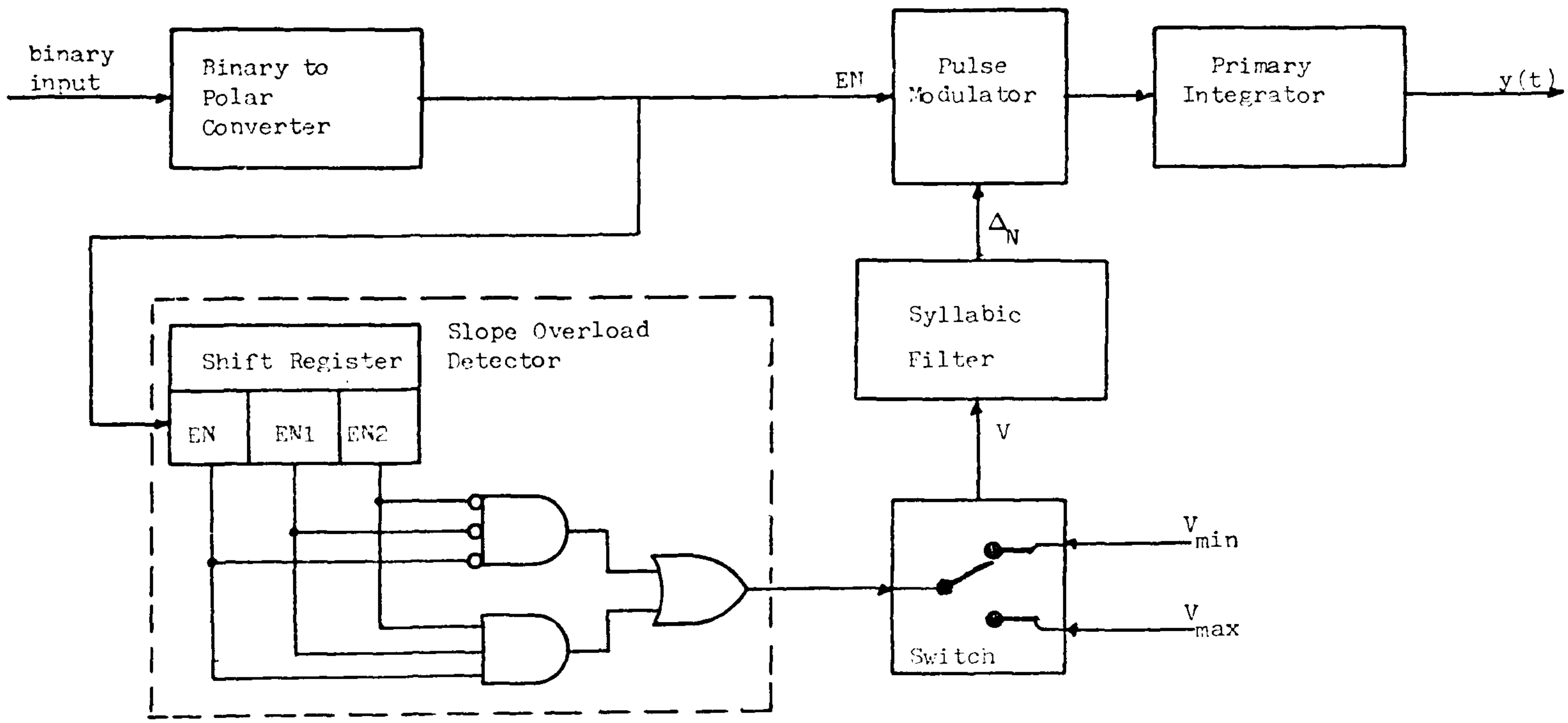


Figure 5. CVSD Decoder Block Diagram

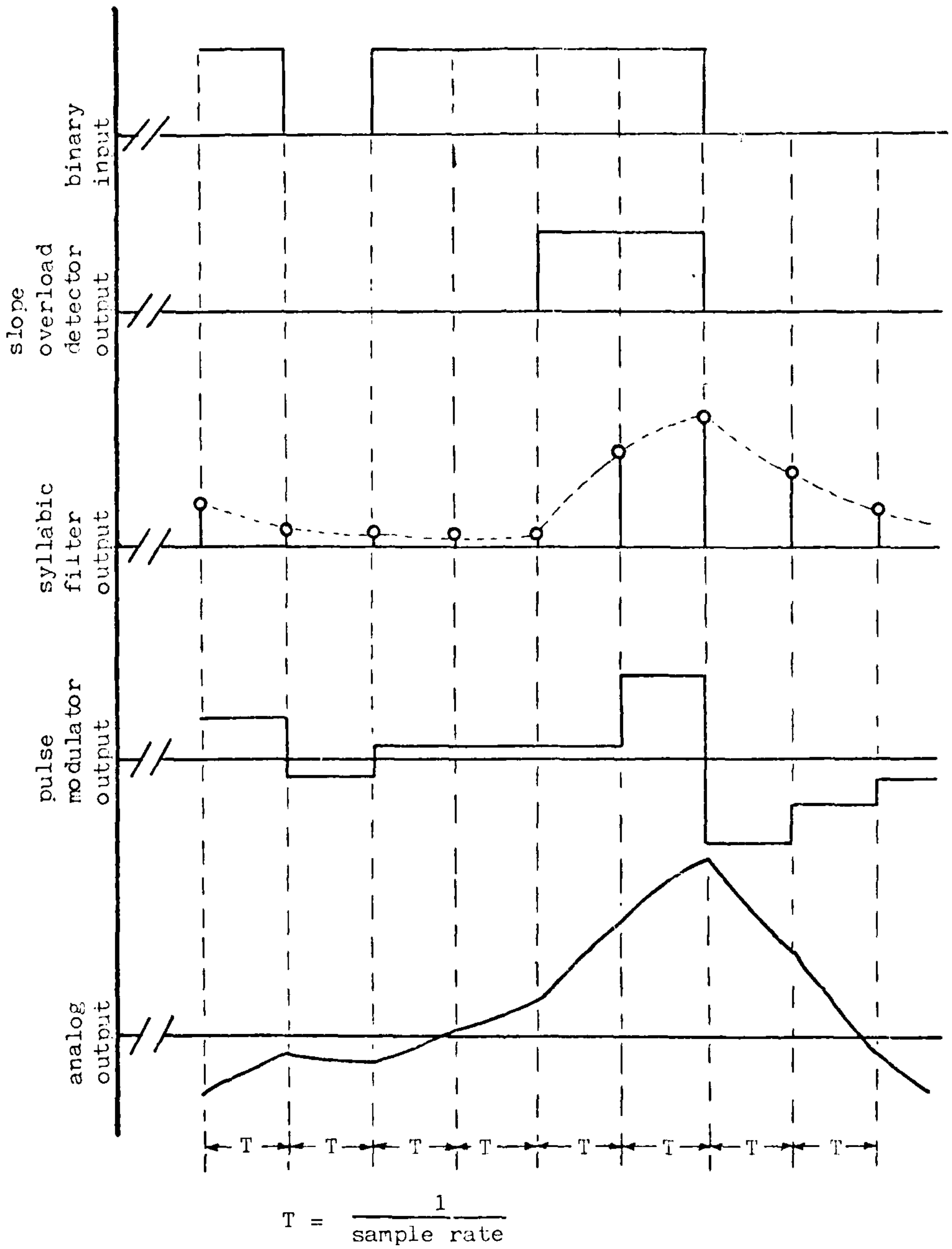


Figure 6. Sample Waveforms at Various Points in the Decoder

$$y(t) = \left[y(t - [N-1]T) \right] e^{-k_1 t} + (1 - e^{-k_1 t}) \frac{a_N}{k_1} \qquad (9)$$

for $(N-1)T < t < NT$

where

$$T = \frac{1}{\text{sample rate}}$$

$$k_1 = 2\pi f_{c1}$$

a_N = the primary integrator input for $(N-1)T < t < NT$

f_{c1} = the pole frequency of the primary integrator in hertz

Analysis of the decoder syllabic filter is identical to that of the encoder syllabic filter and equation (8) also applies to the decoder.

$$\Delta_N = \Delta_{N-1}\beta + (1 - \beta) \frac{V_N}{k_2} \qquad (8)$$

where

$k_2 = \dfrac{2\pi}{t_c}$, Δ_N = the syllabic filter output

$$\beta = \exp\left[-\frac{2\pi T}{t_c} \right]$$

V_N = either V_{max} or V_{min}, the syllabic filter input

<u>Decoder Computer Subroutine</u> Using equations (6) and (8), the decoding subroutine is implemented as shown in the flowchart in figure 7. Equation (9) is not used since the straight line approximation provided by the Calcomp plotter provides a sufficiently accurate representation of the decoder output for this investigation. Except for the elimination of the comparison step used in the encoder subroutine, the decoder subroutine is nearly identical to that of the encoder. All comments applicable to the encoder subroutine are also applicable to the decoder subroutine. The FORTRAN code for the decoder subroutine is attached in Appendix C.

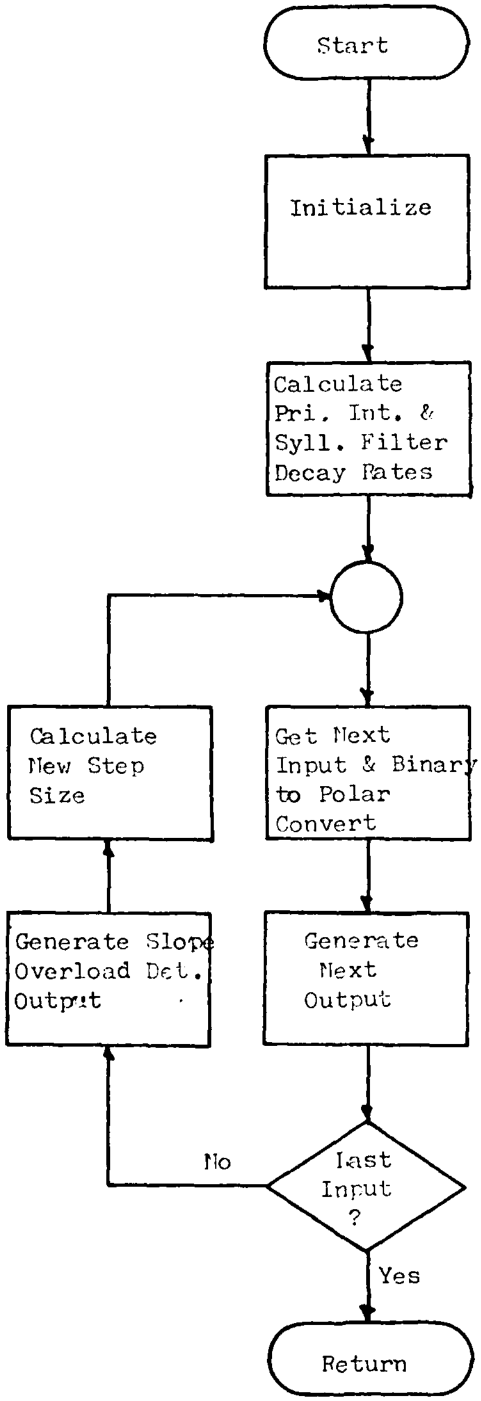


Figure 7. CVSD Decoder Subroutine Flowchart

Encoder and Decoder Parameters From the expressions developed in the preceding sections describing the CVSD encoder and decoder, it can be seen that there are four parameters that determine the characteristics of the encoder and decoder. They are, f_{c1} for the primary integrator, t_c for the syllabic filter, and V_{max} and V_{min} whose value determines the magnitude of the step sizes.

The draft standard specifies the value of f_{c1} explicitly in paragraph 3.2. When the primary integrator consists of a single pole filter, the value of f_{c1} is required to be between 100 and 300 Hz. Other poles and zeros can be added to the primary integrator, in accordance with the draft standard, however, only f_{c1} is required. In this investigation, the single pole version of the primary integrator is used in the CVSD encoder and decoder models.

For the syllabic filter, the draft standard does not specify the value of t_c directly. Instead, t_c is specified in terms of the decoder output signal when a given input is applied to the encoder. When the analog input signal at 800 Hz is stepped from -42 dBm0 to 0 dBm0, the decoder output signal is required to achieve 90% of its final value within 2 to 4 milliseconds after the output signal starts to rise. (NOTE: For this system, the standard specifies the reference test level point to be -4 dBm. So, a -42 dBm0 is actually -46 dBm. All measurements taken in this investigation are stated in dBm0, unless explicitly stated otherwise.)

The values of V_{max} and V_{min} are also not specified directly by the draft standard, but are specified in terms of the syllabic filter output. The syllabic filter output, which has previously been defined as the step size of the encoder and decoder, is required to be linear as a function of the slope overload detector duty cycle. The slope overload detector duty cycle is defined as the ratio of the number of times slope overload is detected to the number of samples in the same period. In paragraph 3.4 of the draft standard, the step size is shown as varying linearly as the duty cycle ranges from 0 to .5. The step size ratio, the ratio of the syllabic filter output when an 800 Hz, 0 dBm0 signal is applied to the encoder input, to the syllabic filter output when the encoder input is grounded is required to be 34 dB ± 2 dB. This specification

in combination with the specifications for f_{c1} and t_c determine the values of V_{max} and V_{min}.

Due to the fact that the parameters interact with each other, the values of t_c, V_{max}, and V_{min} need to be determined recursively. A value of f_{c1} is chosen within the range given by the standard and an estimate of t_c chosen near its expected value. The syllabic filter determines the system response to the amplitude modulation of a voice signal. As the highest frequency in the envelope of the voice signal is generally about 100 Hz, t_c is estimated to be the reciprocal of this frequency or .01. A nominal step size ratio is given by the draft standard to be 34 dB. These three parameters are used to calculate the values of V_{max} and V_{min}. Figure 8 is the flowchart of the program that calculates these values using the subroutines shown in figures 9 and 10, then plots the resulting syllabic filter output as a function of slope overload detector duty cycle.

Initially, estimated values of V_{max} and V_{min} are used and the slope overload detector duty cycle and system step size ratio calculated when an 800 Hz, 0 dBm0 test signal is input to the CVSD encoder. If the calculated values are not within the tolerances specified, V_{max} and V_{min} are adjusted and the calculations repeated. This process is continued until values of V_{max} and V_{min} are determined that produce a slope overload detector duty cycle of .5 ± 1%, and a step size ratio within .01% of the input value.

After determining the values of V_{max} and V_{min}, the entire CVSD system is tested to determine if the rise time requirement is met using the parameters that have been calculated. The flowchart of the test program is shown in figure 11. To determine the system rise time, a test signal consisting of alternate series of 500 samples of an 800 Hz, -42 dBm0 sine wave and 500 samples of the 800 Hz sine wave at 0 dBm0. The initial series at -42 dBm0 initialize the storage elements of the slope overload detectors in both the encoder and decoder and get the system into an initial steady-state condition. After processing the test signal through the system, the output signal is plotted in the vicinity around one of the steps in input signal power. The system rise time is then determined graphically. Figure 12 shows a sample output from this program for both the 16 and 32 kb/s sample rates. This test was performed with f_{c1} = 100 Hz, step size ratio = 34 dB, t_c = .01 for the 16 kb/s sample rate, and

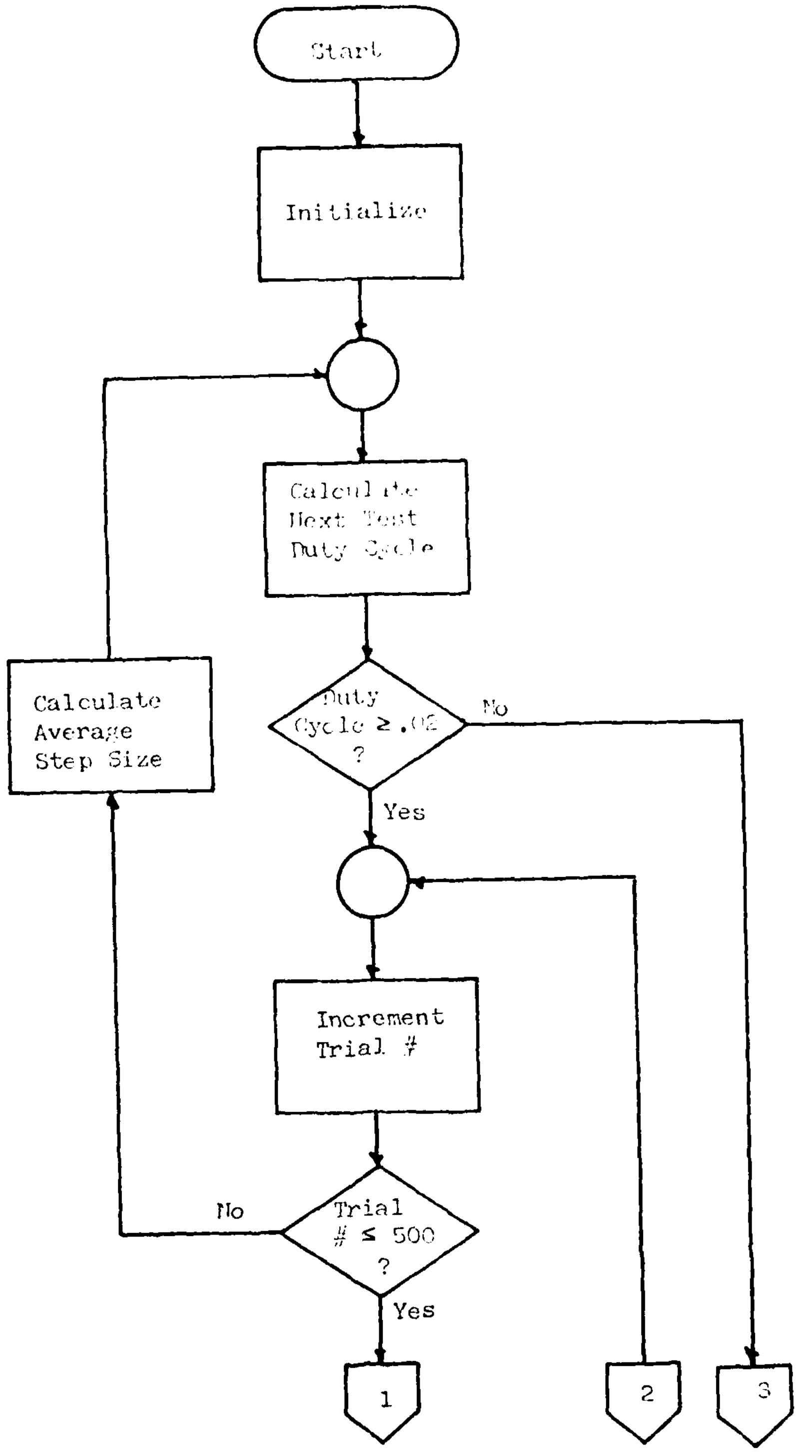


Figure 8. Syllabic Filter Output Amplitude Response Test Program Flowchart (STEPS?)

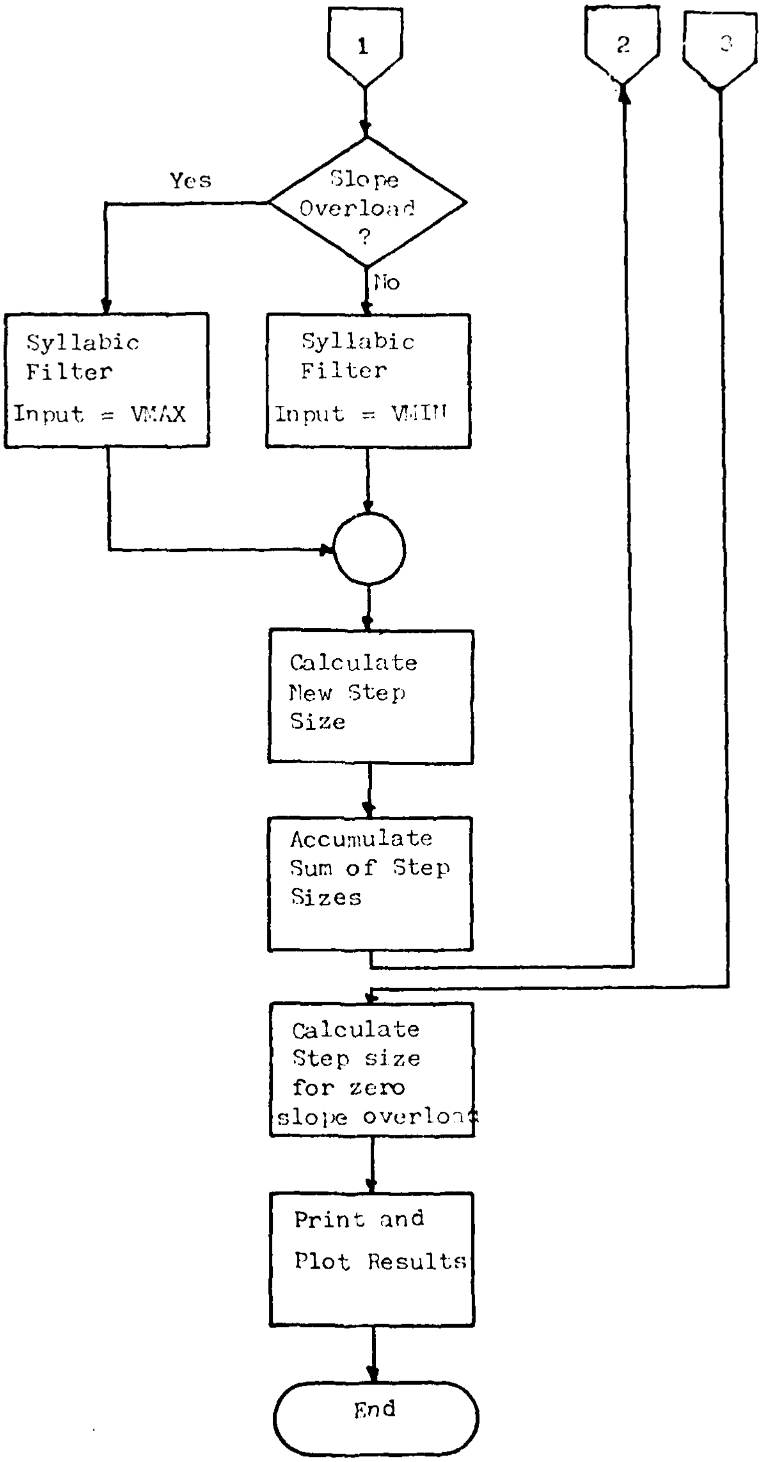


Figure 8. (continued)

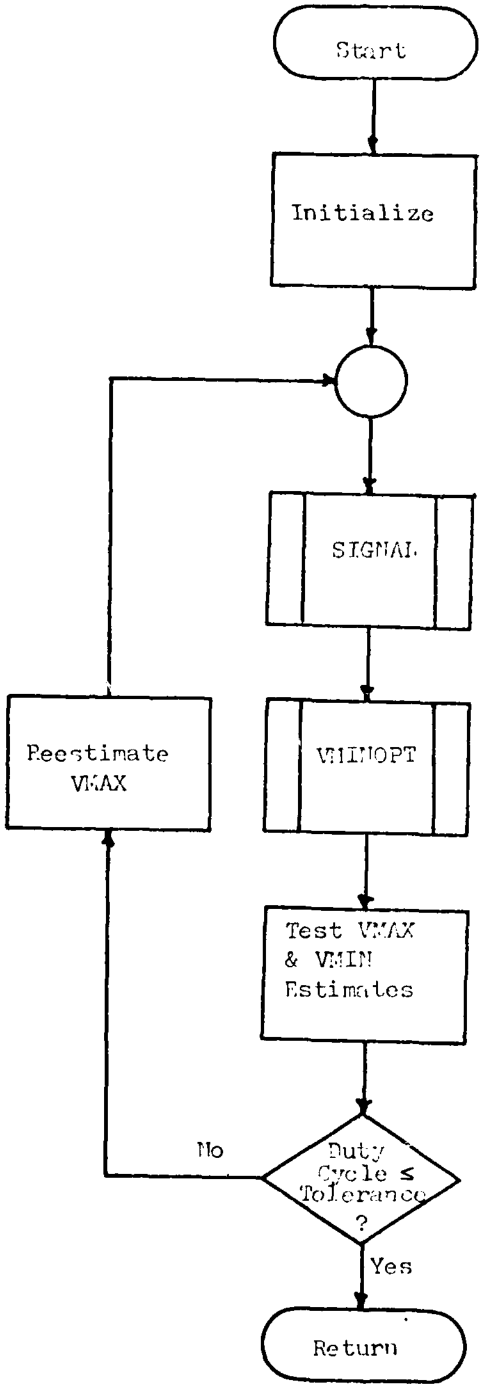


Figure 9. CVSD System Parameter Calculation Subroutine Flowchart (VMAXOPT)

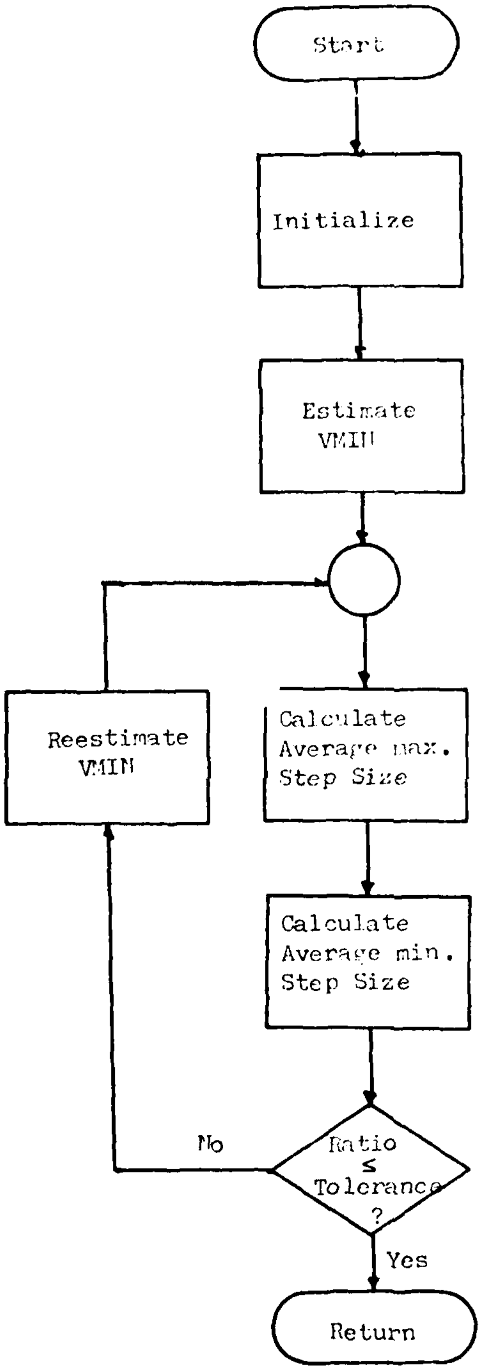


Figure 10. CVSD System Parameter Calculation Subroutine Flowchart (VMINOPT)

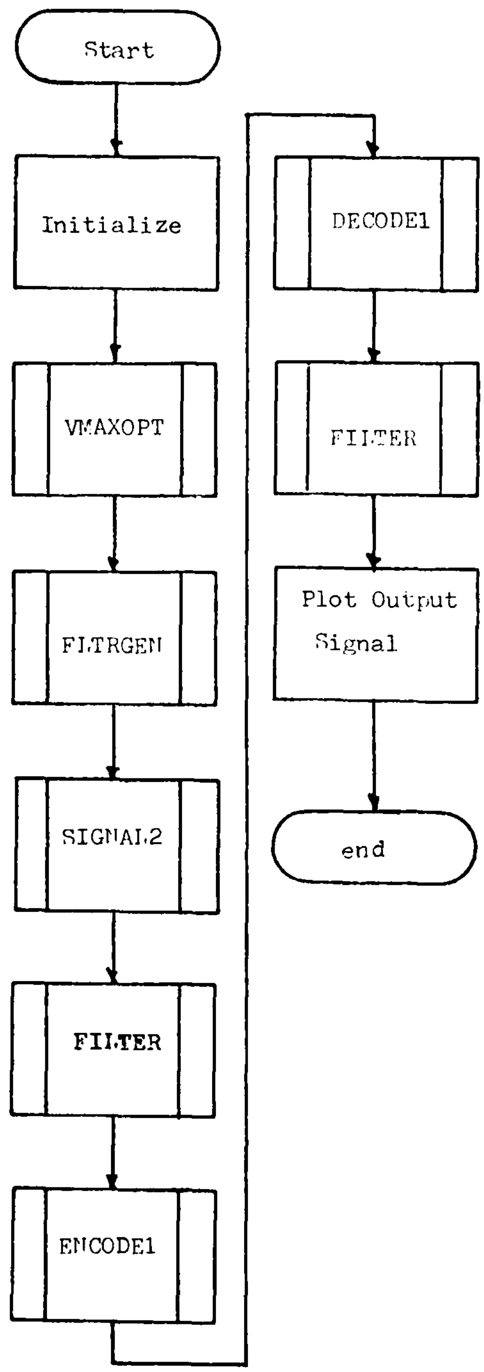


Figure 11. CVSD System Step Response Program Flowchart (PULSE)

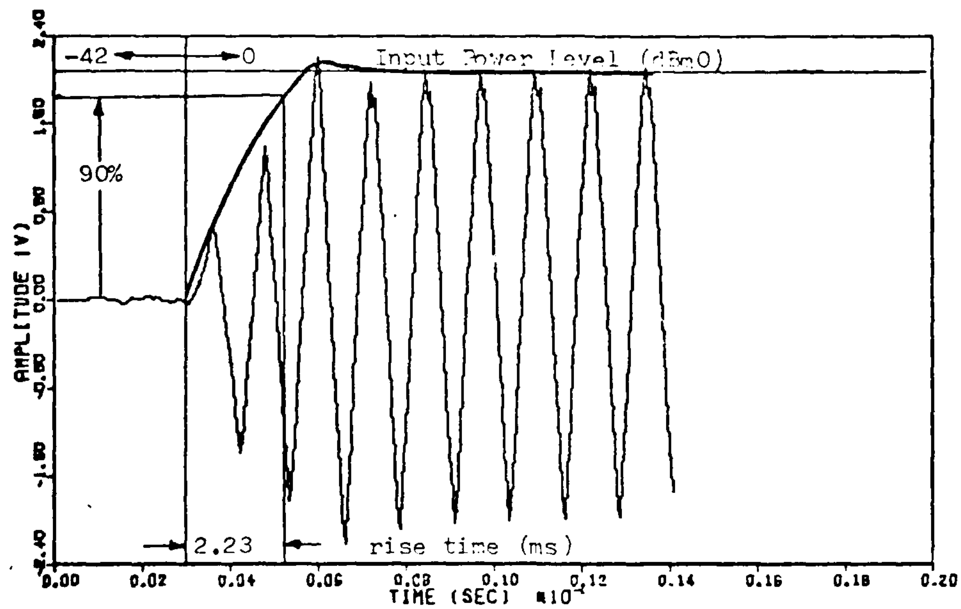


Figure 12a. CVSD System Response to an 800 Hz Step Signal at 16 kb/s Sample Rate

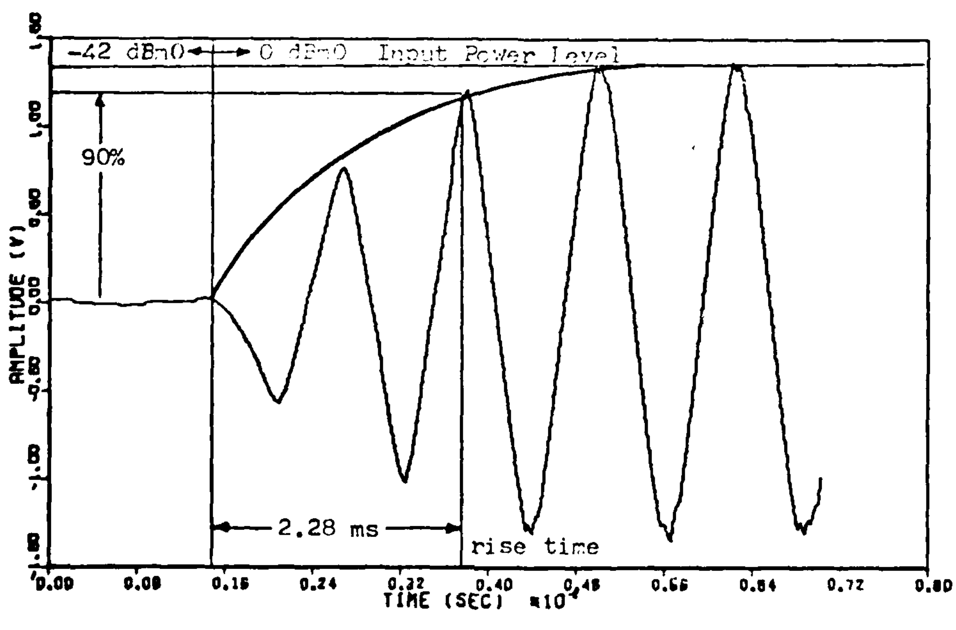


Figure 12b. CVSD System Response to an 800 Hz Step Signal at 32 kb/s Sample Rate

t_c = .02 for the 32 kb/s test. Table I summarizes the parameters and the ranges of each that will still result in a system that complies with the performance criteria set by the draft standard.

TABLE I

PARAMETER SUMMARY

Parameter		Sample Rate	
		16 kb/s	32 kb/s
Primary Integrator	$- f_{c1}$	100 - 300 Hz	
Syllabic Filter	$- t_c$	.01 - .015	.02 - .025
Step Size Ratio		34 dB ± 2 dB	

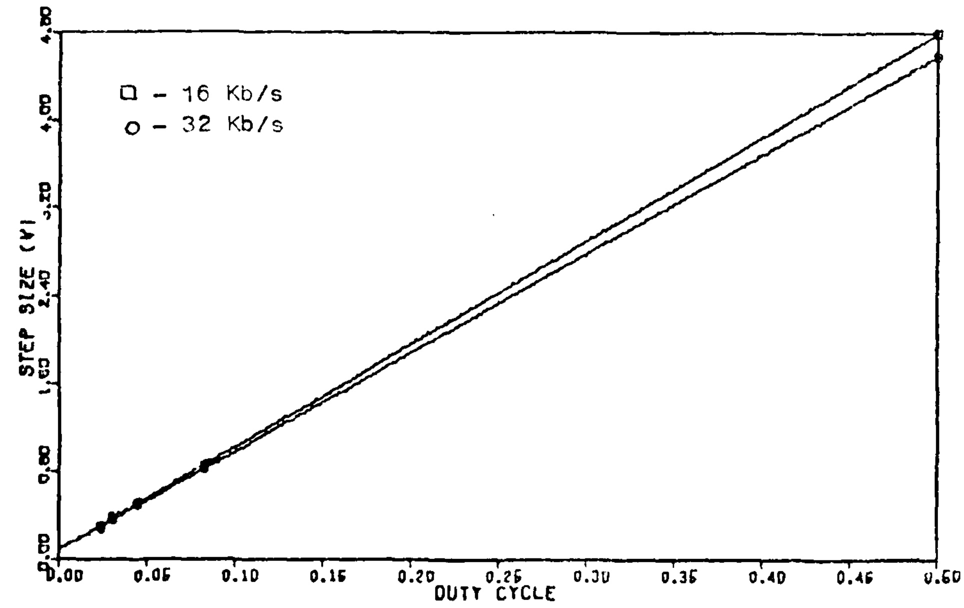


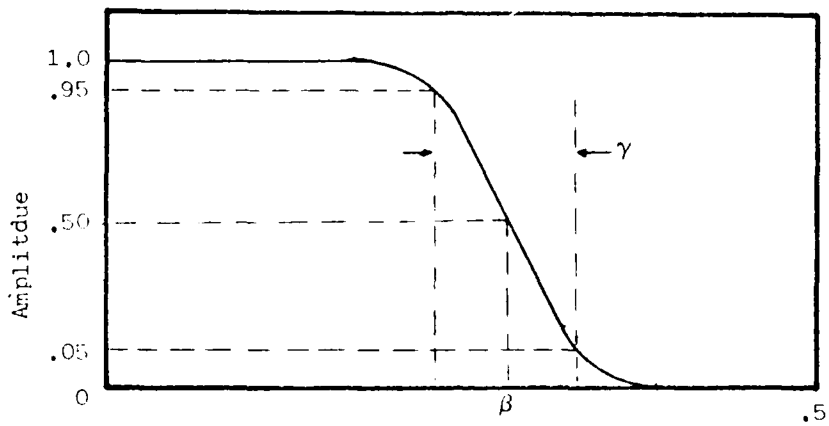
Figure 13. Syllabic Filter Output (step size) as a Function of Slope Overload Detector Duty Cycle for both 16 and 32 kb/s Sample Rates

Input and Output Low-Pass Filters The last of the major components making up the CVSD analog/digital conversion system are the input and output filters. These filters are used to limit both the input and output signal spectrum to the voice band frequencies only. For telephonic communications, the voice band is generally considered to be those frequencies less than 3600 Hz. For optimal system performance, these filters should have a very sharp cut-off and high loss characteristics in the stop band.

The purpose of the input filter is to limit the input signal spectrum to prevent aliasing due to the sampling process. When the input signal is sampled, in addition to the input spectrum, the output spectrum also contains sum and difference frequency components centered around the sample frequency. If the input spectrum were to contain frequencies very much larger than the desired spectrum, aliasing or interference would occur when the difference frequencies fell into the baseband spectrum. For this model, the input filter is considered to be an ideal low-pass filter. The test signal generator output spectrum is limited to the voice band frequencies only, with no components falling outside that range. This simulates a low-pass filter with zero insertion loss in the pass band and infinite loss in the stop band.

The function of the output filter is also to limit the signal to the voice band, however, in this case, the components outside the original input spectrum are produced by the non-linearities of the processing system. The output filter smooths the signals and eliminates the harmonic components above the voice band. This filter may have the same characteristics as the input filter or may have a narrower pass band to improve performance. The filter chosen for this model is a maximally flat, linear phase symmetrical finite impulse response (FIR) filter. The model of this filter was developed by J.F. Kaiser of the Digital System Research Department of the Bell Laboratories. Reference 1 provides more complete documentation of the filter model. There are two parameters that define the response of the filter, beta and gamma. Figure 14 shows the response of the filter generated by this program and where beta and gamma are defined. Beta is the normalized center frequency of the transition region and gamma is the normalized width of the region. Normalization is with respect to the sample rate. This filter was chosen for its flat

response in the pass band in order to minimize disturbance of the CVSD encoder/decoder response, since those are the primary system components under investigation. The parameters chosen for the filter are β = .1875 and γ = .1 for the 16 kb/s sample rate and β = .1 and γ = .1 for the 32 kb/s sample rate. Table II lists the filter coefficients generated by the program and figure 15 shows the frequency response for both filters.



Normalized Frequency

Figure 14. Maximally Flat FIR Filter Response Characteristic and Parameter Definition

The center of the transition region for the 16 kb/s filter is 3 kHz and 3.2 kHz for the 32 kb/s filter. As can be seen, the maximally flat characteristic is achieved at the expense of stop band loss. However, it will be shown in the performance results that the system performance meets most of the criteria specified by the draft standard in spite of the poor filter performance.

Filter Subroutines The filter program developed by J.F. Kaiser is used to generate the FIR filter coefficients, however, it has been modified to be a subroutine that returns the coefficient values to the calling program instead of printing them out. These coefficients are produced by FLTRGEN then used by subroutine FILTER to actually filter the signal input to the filter. The maximum number of coefficients that can be produced by FLTRGEN without program modification is 200. Subroutine FILTER delays the output signal by 200 sample periods so that it has at least

TABLE II

FIR FILTER COEFFICIENTS

16 kb/s sample rate		32 kb/s sample rate	
B(1) =	.3740[illegible]72926	B(1) =	.1945846232
B(2) =	.2302383251	B(2) =	.13[illegible]3456867
B(3) =	.1[illegible]73955562	B(3) =	.1427597[illegible]4[illegible]
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B(5) =	-.[illegible]653779172[illegible]	B(5) =	.0428271121
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B(9) =	.0[illegible]7836[illegible]423	B(9) =	-.0137595193
B(10) =	-.0123795254	B(10) =	-.01[illegible]5310321
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B(15) =	-.[illegible]0124221[illegible]27	B(15) =	.[illegible]02173[illegible]2[illegible][illegible]
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B(20) =	-.[illegible]1[illegible]0[illegible]52624	B(20) =	-.[illegible]0[illegible]2[illegible]4395[illegible]
B(21) =	-.[illegible]001[illegible]31339	B(21) =	-.[illegible]0[illegible]1382753
B(22) =	-.[illegible]0[illegible]149[illegible][illegible]61	B(22) =	-.[illegible]0[illegible]07881[illegible]7
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B(35) =	-.[illegible][illegible][illegible][illegible][illegible][illegible][illegible]4	B(35) =	-.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(36) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible]2	B(36) =	-.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(37) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible]1	B(37) =	-.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(38) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]	B(38) =	-.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(39) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]	B(39) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(40) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]	B(40) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(41) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]	B(41) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]
B(42) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]	B(42) =	.[illegible][illegible][illegible][illegible][illegible][illegible][illegible][illegible]

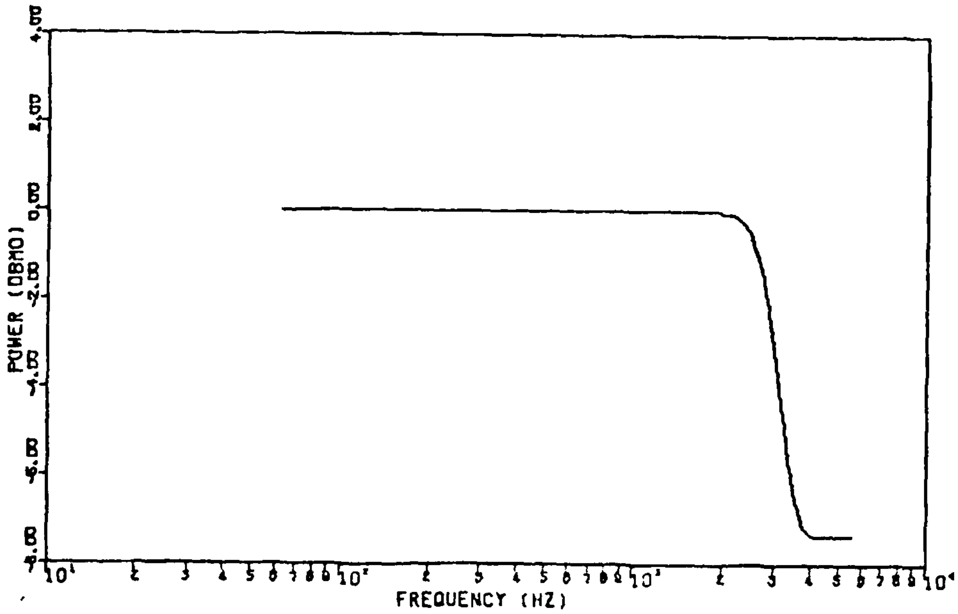

Figure 15a. CVSD System Model Output Filter Response
for 16 kb/s Sample Rate

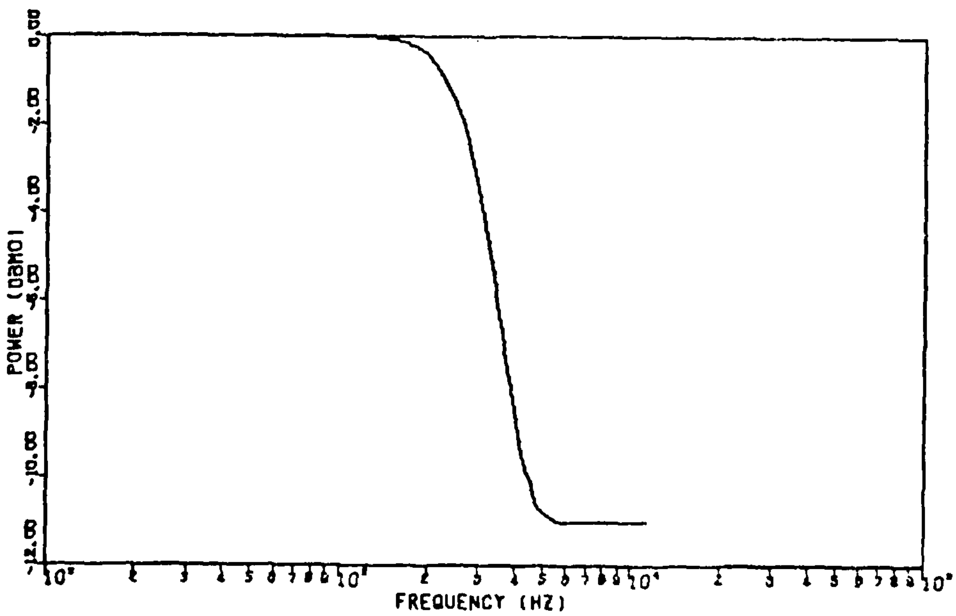

Figure 15b. CVSD System Model Output Filter Response
for 32 kb/s Sample Rate

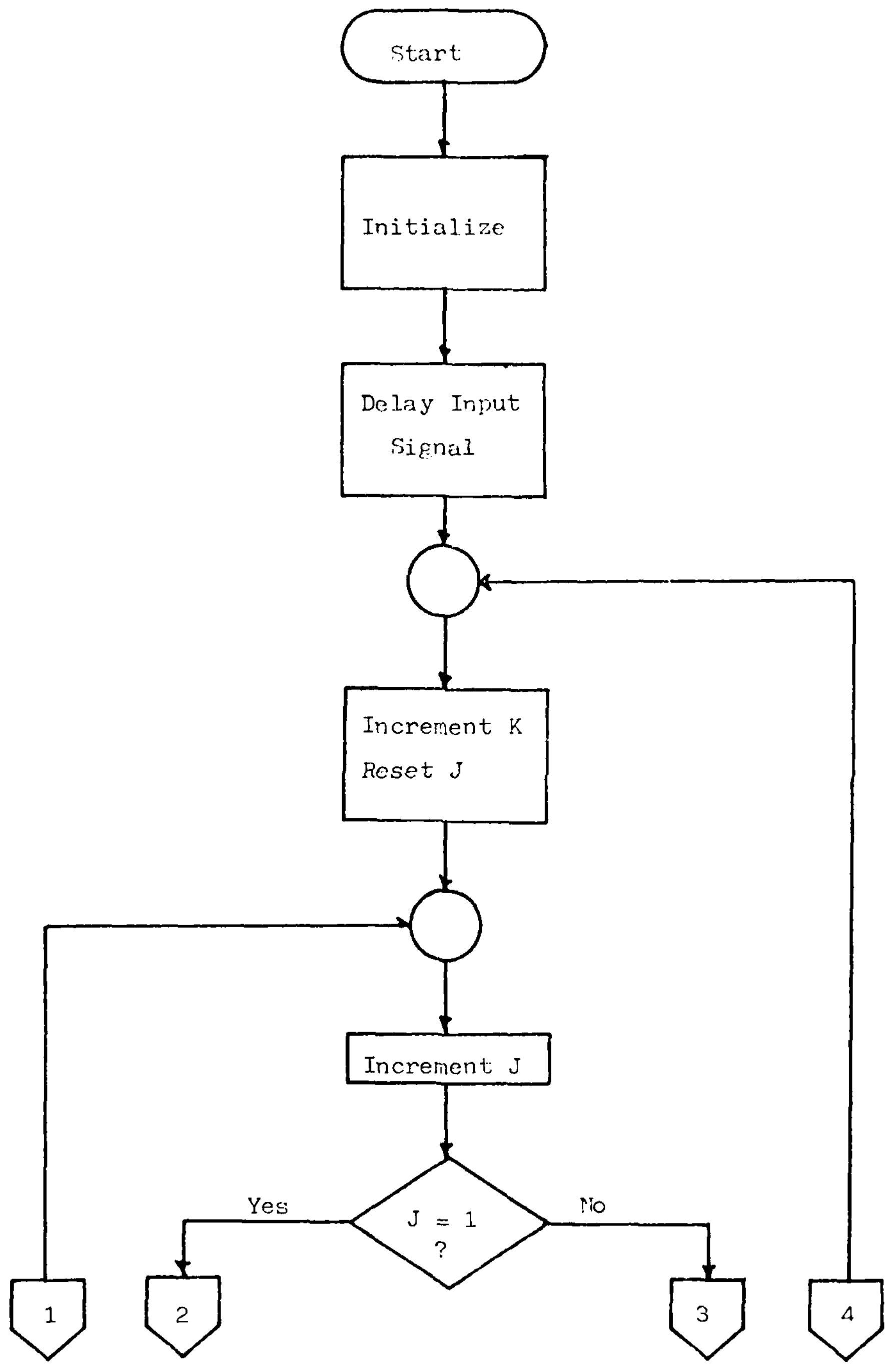

Figure 16. Signal Filtering Subroutine Flowchart (FILTER)

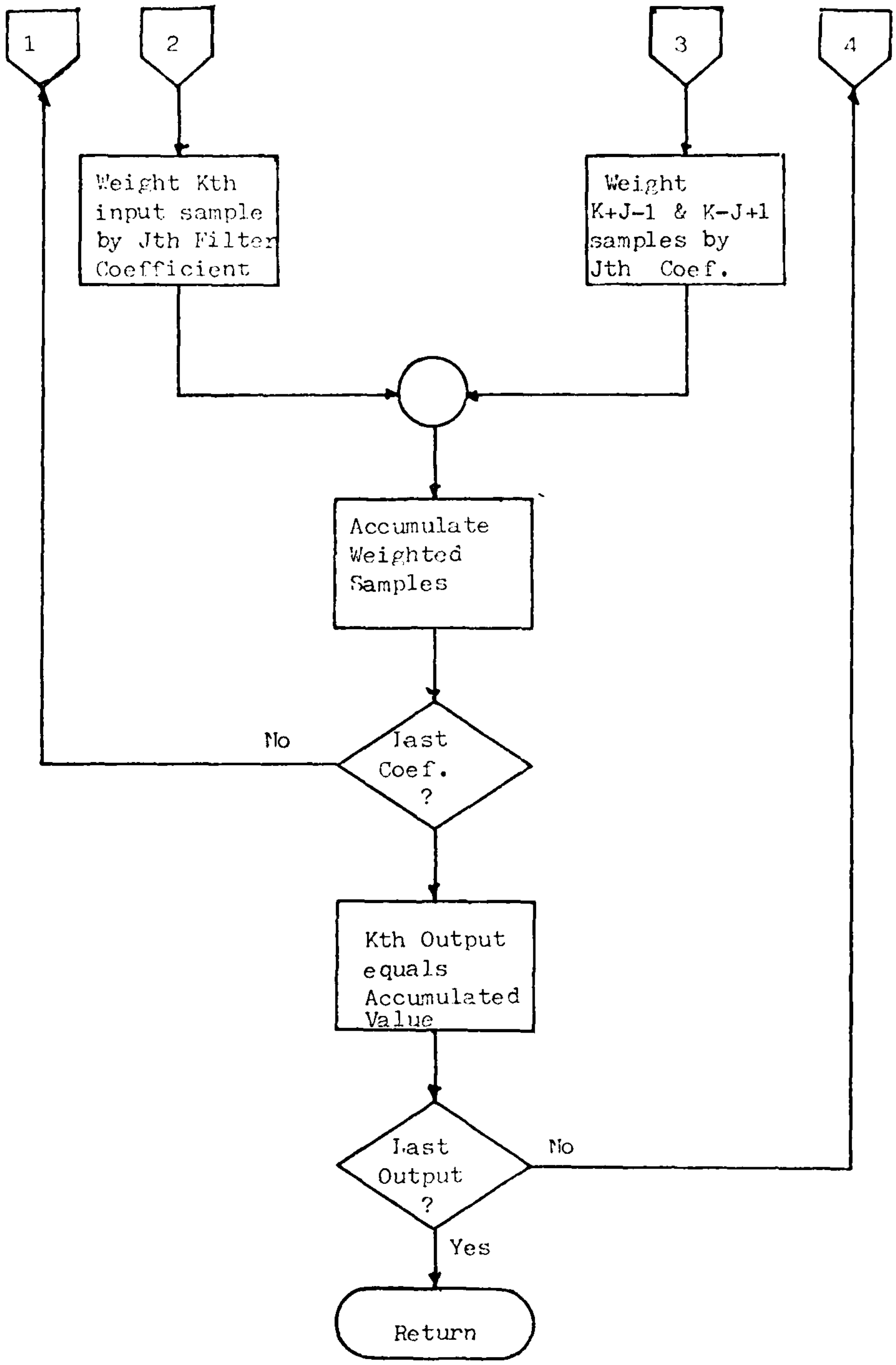

Figure 16. (continued)

200 input signal samples can be used by the filtering algorithm. Equation (10) shows the filtering expression implemented by the FILTER subroutine.

$$y_n = B_1 x_n + \sum_{i=2}^{NP} B_i (x_{n+i-1} + x_{n-i+1}) \qquad (10)$$

where

y_n = the nth output sample

x_n = the nth input sample

B_i = the ith FIR filter coefficient

NP = the number of filter coefficients

III. Performance Tests

A model of the continuously variable slope delta analog to digital conversion system is constructed from the component models described in the previous sections. Figure 17 shows the test configuration simulated by the computer model used in this investigation. The system under test is shown in figure 1. In this simulation, the test signal generator is a subroutine that generates samples of a sinusoidal signal that can be composed of up to two frequency components at individually specified amplitudes. The standard test signal used in the performance tests is an 800 Hz sine wave at -20 dBm0, unless otherwise stated. As previously indicated, the reference signal level is -4 dBm. All power measurements are made relative to this level. The test signal is generated as an array of 5000 samples for most of the tests performed. This array is then processed through the system, the output array of each system component becoming the input array of the next. The final system output signal is then processed to determine the various signal characteristics. System performance is measured in terms of the commonly used voice frequency tests as, idle channel noise, total harmonic distortion, intermodulation distortion, signal-to-noise ratio, and frequency response. These tests are first performed with the CVSD encoder and decoder parameters matched, then performed again with various combinations of encoder and decoder parameters to show how system performance degrades under mismatched conditions.

Idle Channel Noise Test Idle channel noise is measure of the basic amount of noise that the processing system adds to the output signal. The output signal power is measured while the system input is grounded. Any non-zero power measured is the idle channel noise. Before measuring the idle channel noise, however, the system insertion loss is first set so that the standard test signal experiences no change in power after being processed through the system. Idle channel noise is then measured as,

$$ICN = \frac{1}{N} \sum_{n=1}^{N} y_n^2 \qquad (11)$$

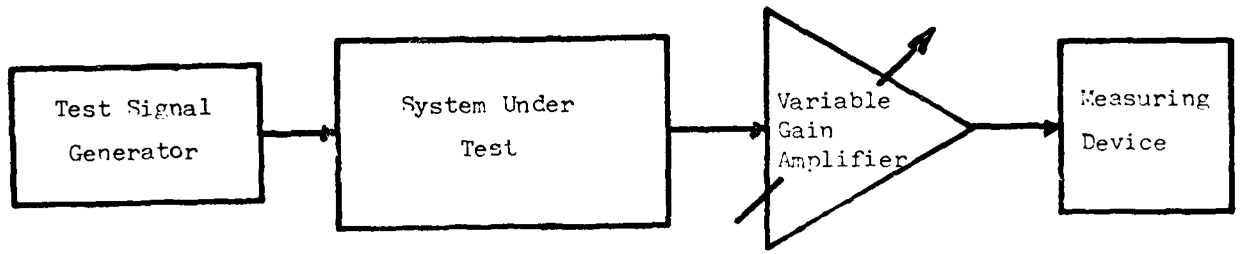

Figure 17. Simulated System Test Configuration

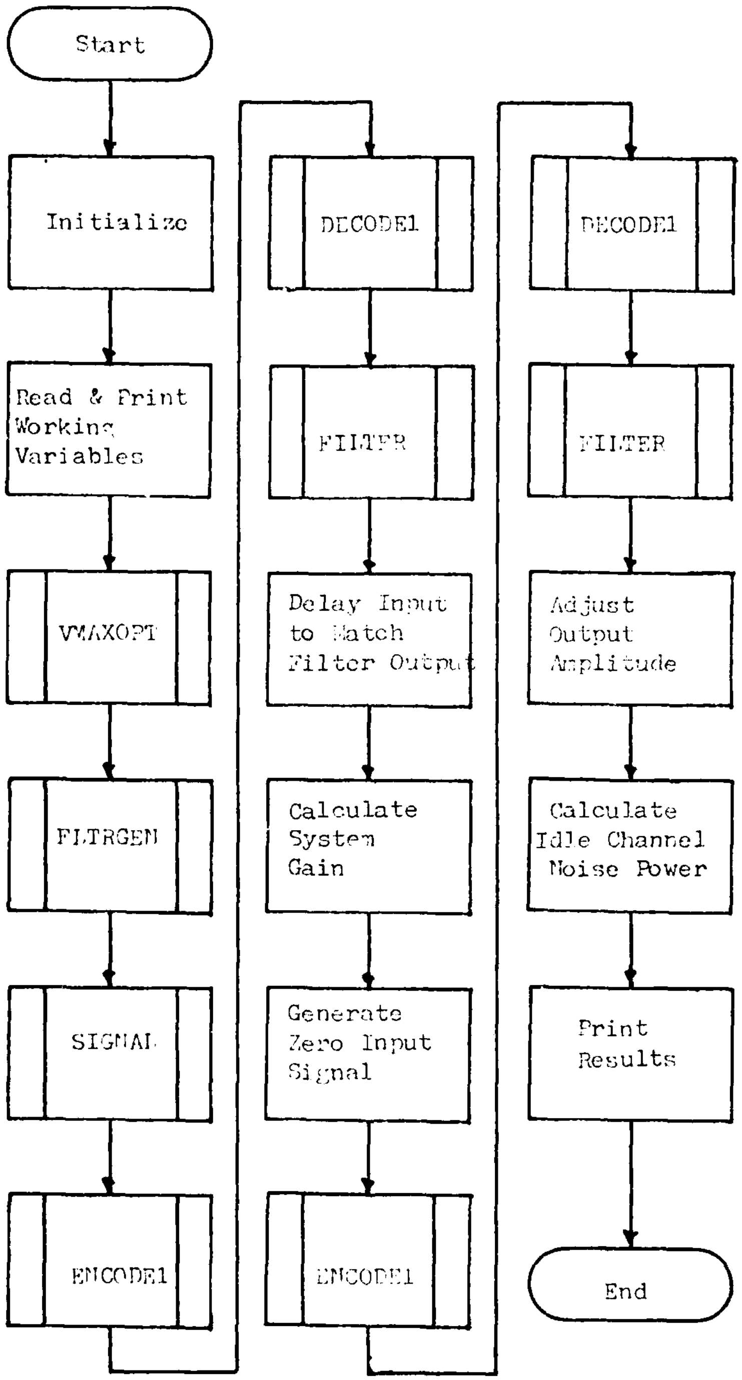

Figure 18. Idle Channel Noise Program Flowchart

where

y_n = the output signal amplitude

N = the number of samples

Figure 18 is the flowchart of the idle channel noise test used to measure the CVSD system performance.

Total Harmonic Distortion Test Total harmonic distortion is one of the measures of system non-linearity. The CCITT procedure for measuring total harmonic distortion is to input a single frequency test signal near the center of the system's pass band and measure the magnitude of the harmonic components in the output spectrum. Total harmonic distortion is then calculated by,

$$THD = \frac{\sqrt{E_2^2 + E_3^2 + \cdots + E_N^2}}{E_1} \times 100\% \qquad (12)$$

where

$E_2, E_3, \cdots, E_N$ = the RMS voltages of the harmonic signal components in the output spectrum

E_1 = the RMS voltage of the primary signal component in the output spectrum

N = the largest harmonic within the system pass band

Figure 19 is the flowchart of the computer program used to calculate total harmonic distortion. After processing the single frequency sine wave test signal through the CVSD system, the output signal spectrum is calculated using a fast fourier transform (FFT). Due to the limitations of the FFT, the standard test signal is not used, instead, a 1000 Hz signal at -20 dBm0 is used. The FFT procedure can only measure signal components at multiples of the minimum frequency resolution which is determined by the number of samples in the FFT window. In this case, the window length was specified to be 256 samples, which allowed a frequency resolution of 62.5 Hz at the 16 kb/s sample rate and 125 Hz at the 32 kb/s sample rate. A 1000 Hz test signal was chosen as being both compatible with the FFT and a commonly used test signal in voice frequency measurements.

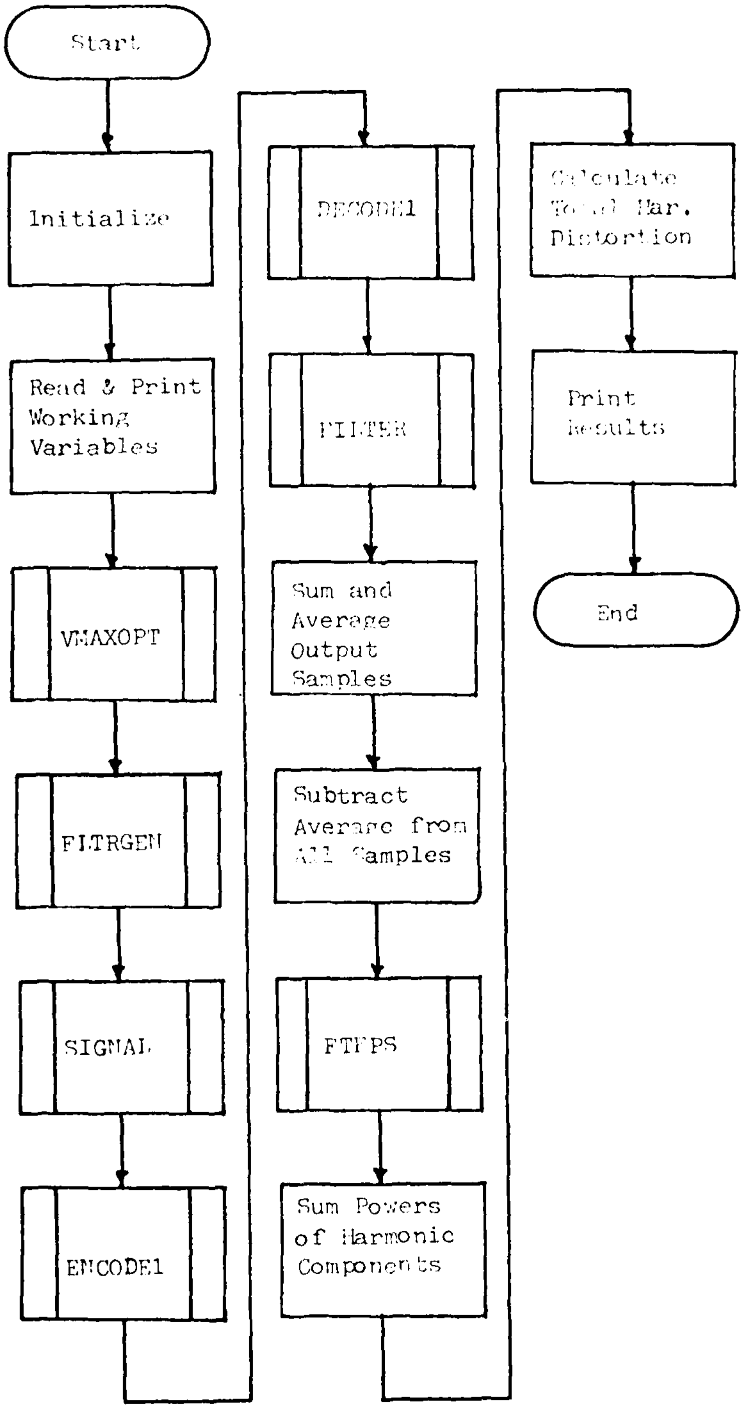

Figure 19. Total Harmonic Distortion Test Program Flowchart (THD)

Intermodulation Distortion The total harmonic distortion test often does not give a complete idea of the system response non-linearities. Intermodulation distortion is another measure of non-linearity used in voice frequency system. The CCITT procedure of measuring intermodulation distortion is to input a composite test signal made of of two sinusoidal signals of equal amplitude. The frequencies of the two signals are separated by an amount that the difference frequency is within the pass band of the system. Intermodulation distortion is then calculated by,

$$\text{INTERMOD} = \frac{E_{dif}}{\sqrt{E_1^2 + E_2^2}} \times 100 \qquad (13)$$

where

E_{dif} = the RMS voltage of the difference frequency component in the output spectrum

E_1 = the RMS voltage of the first frequency [illegible] output spectrum

E_2 = the RMS voltage of the second frequen[illegible] output spectrum

Figure 20 shows the flowchart of the program used to [illegible] modulation distortion for the CVSD system. The [illegible] similar to the total harmonic distortion program [illegible] tral components used from the FFT output are the [illegible] and the difference frequency. The test signal used in the [illegible] sists of 750 Hz and 1000 Hz components at -[illegible] dBm.

Signal-to-Noise Ratio Measurement The signal-to-noise ratio [illegible] measure of how accurately the system being characterized [illegible] input signal. A test signal is process through the system [illegible] ing output signal compared to the input signal after compensat[illegible] the system insertion loss and signal delay. SNR is then calculated by,

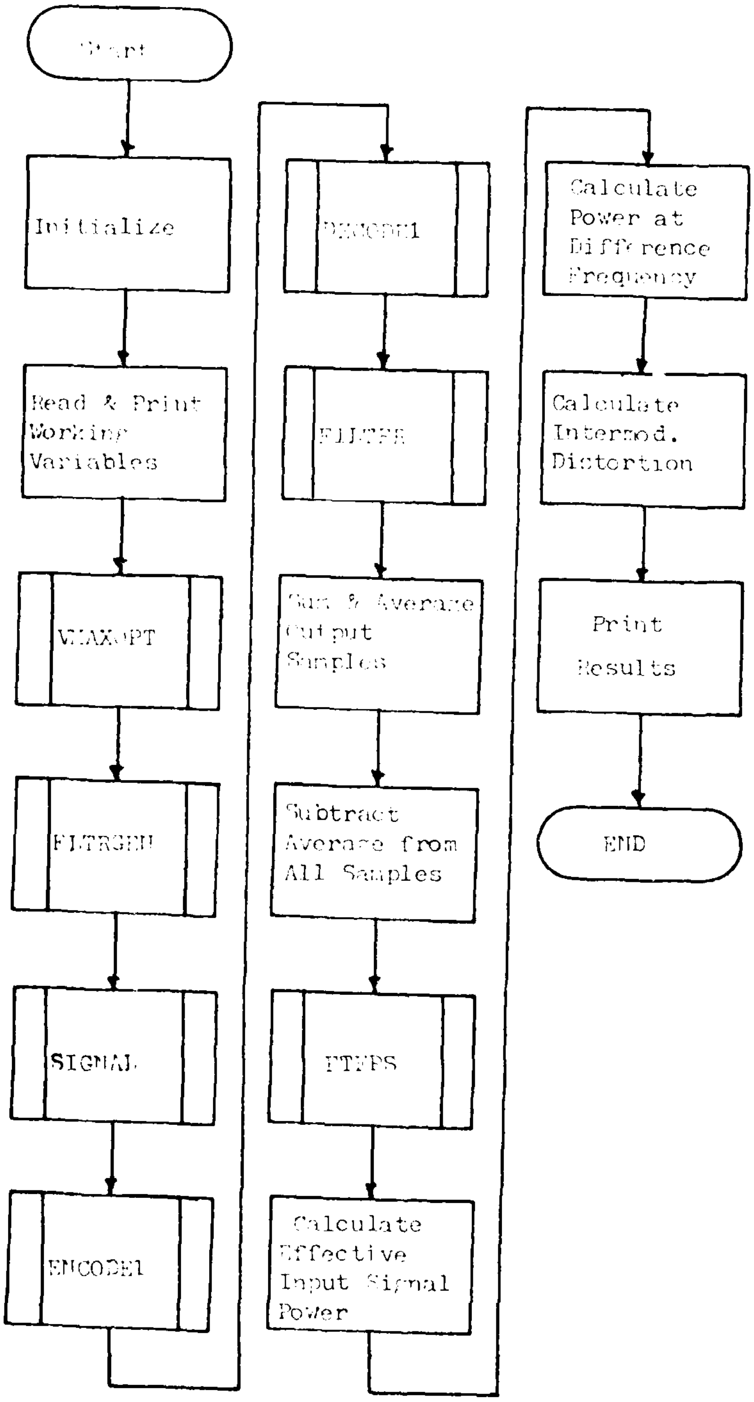

Figure 20. Intermodulation Distortion Test Program Flowchart (INTERMD)

$$SNR = \frac{\sum_{n=1}^{N} (y_n - x_n)^2}{\sum_{n=1}^{N} x_n^2} \qquad (14)$$

where

x_n = the nth input signal sample

y_n = the nth output signal sample

N = the total number of samples

Figure 21 shows the flowchart for the signal-to-noise program used to characterize the CVSD system performance. The standard 800 Hz test signal is used to perform the initial system characterization.

Frequency Response Measurement Two methods of performing frequency response measurement are used in this investigation. The first is flat weighted measurement which is used to determine the frequency response of the entire CVSD system sine this is the method for which the draft standard specifies performance criteria. A second method is the frequency selective measurement of the response characteristics. This method is used to investigate the frequency response of the CVSD encoder and decoder only.

Flat weighted frequency response measurement is performed by inputting a single frequency sine wave test signal at a constant amplitude, then measuring the system output signal power. The output signal includes components at frequencies other than the test signal frequency, however, the power of the entire composite signal is measured without filtering. The measured gain variations are then plotted and scaled such that the 800 Hz measurement is 0 dB. The flowchart of the program using this procedure is shown in figure 22.

Frequency selective measurement of frequency response uses the same test procedure except only the magnitude of the output signal component at the test frequency is measured. The other components of the composite output signal are not included in this measurement. The measurements are then scaled and plotted such that the 1000 Hz measurement is 0 dB. Figure 23 is the flowchart of the program to perform the frequency selective measurements. The 1000 Hz measurement is used as the

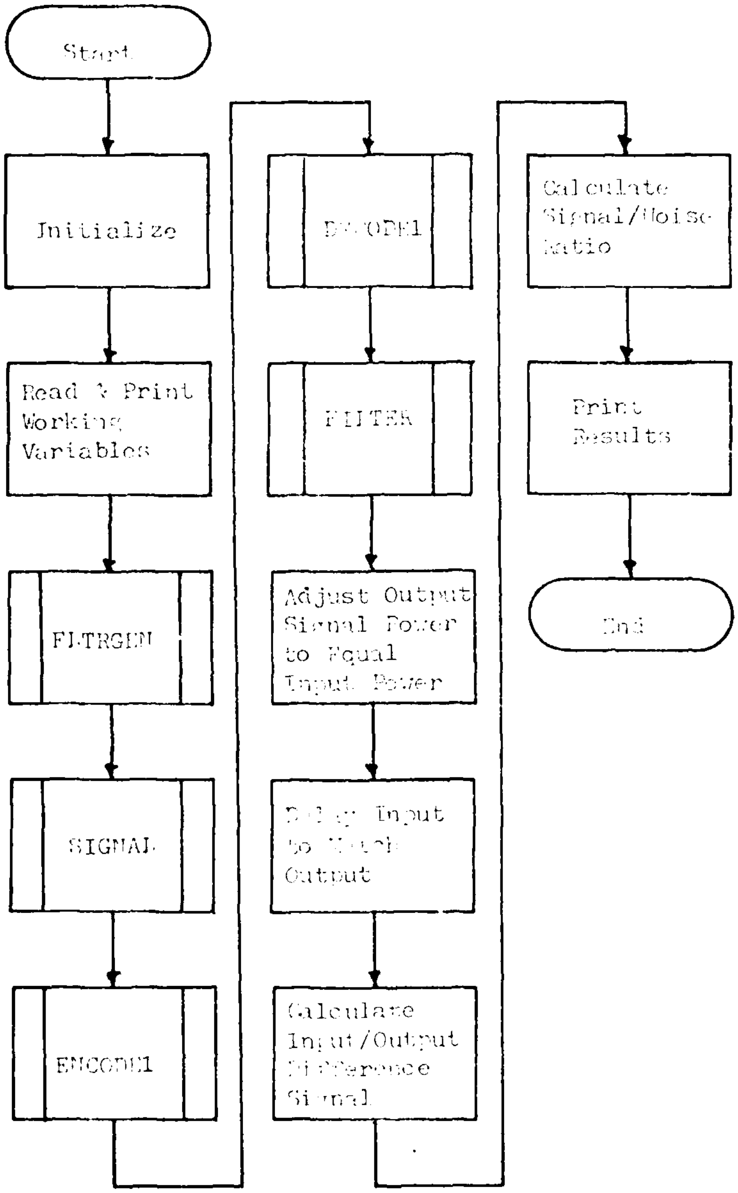

Figure 21. Signal-to-Noise Measurement Program Flowchart

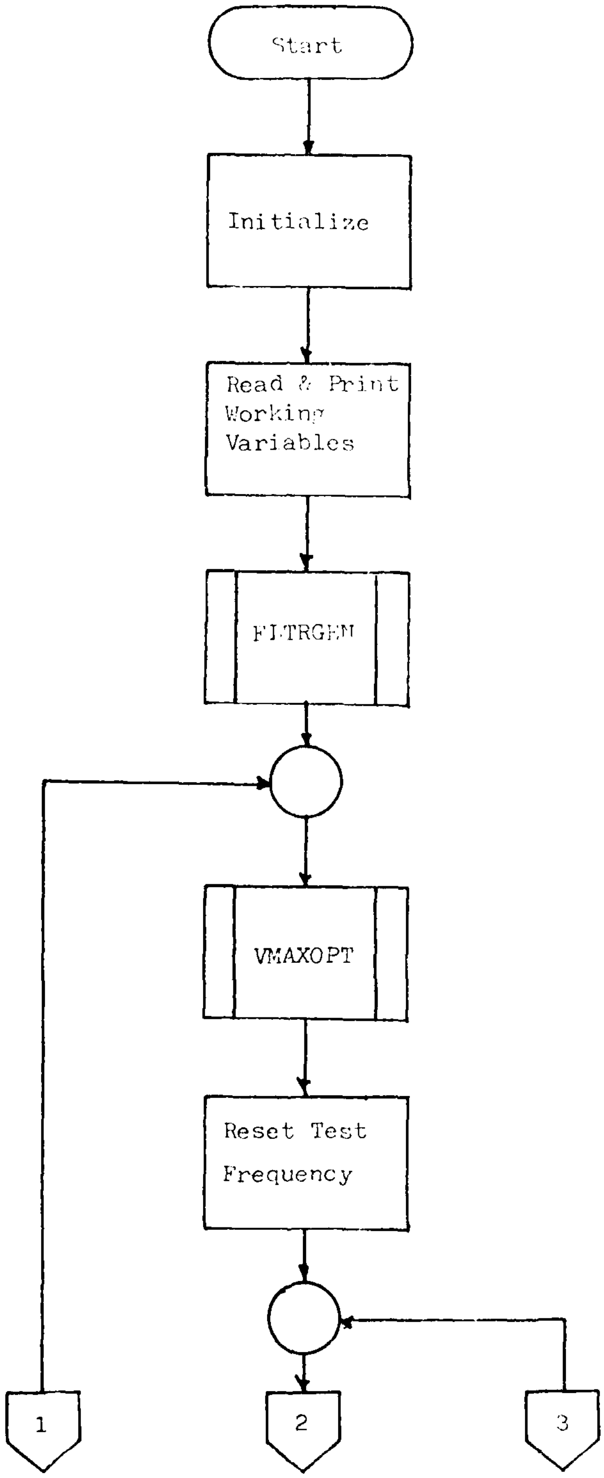

Figure 22. Flowchart of Flat Weighted Measurement of Frequency Response Program (PGAIN)

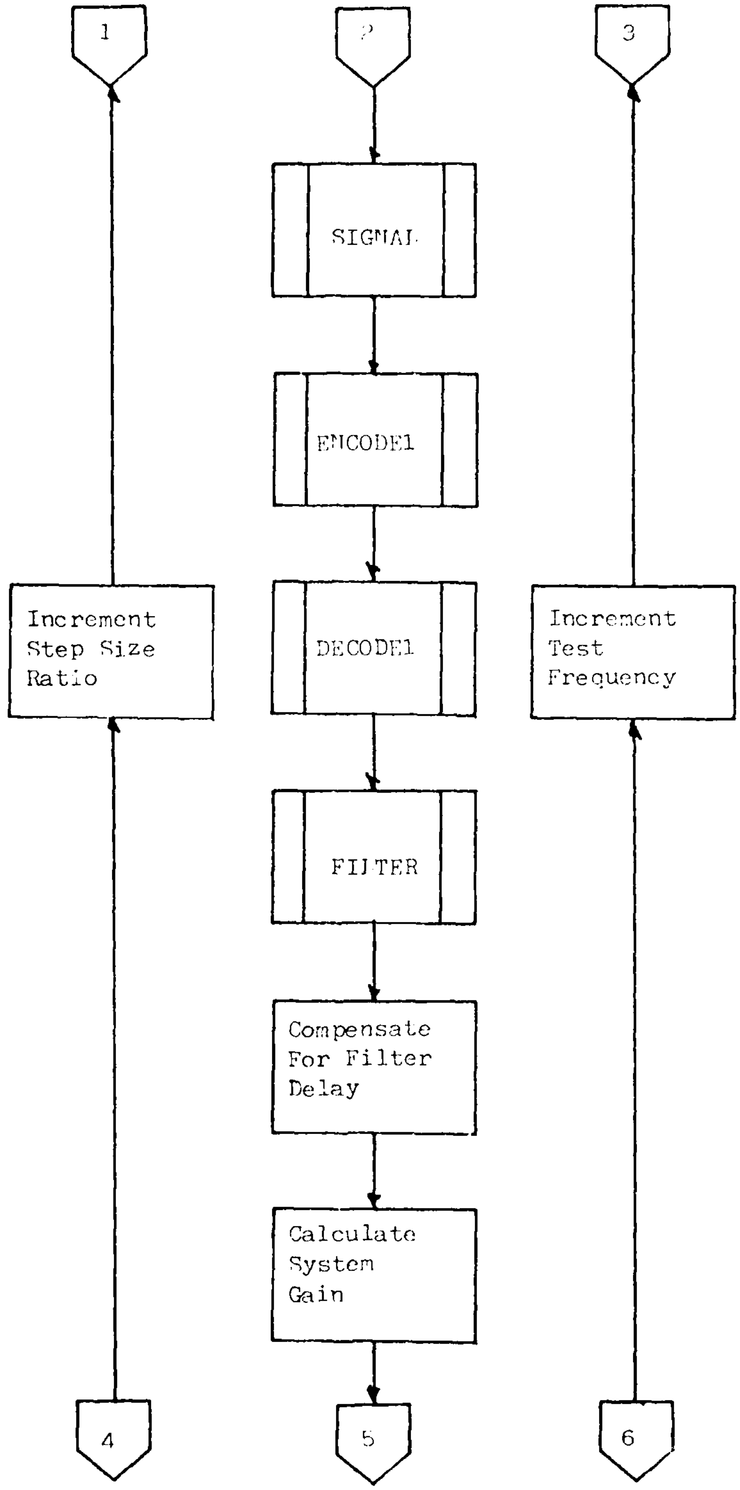

Figure 22. (continued) Program DGAIN Flowchart

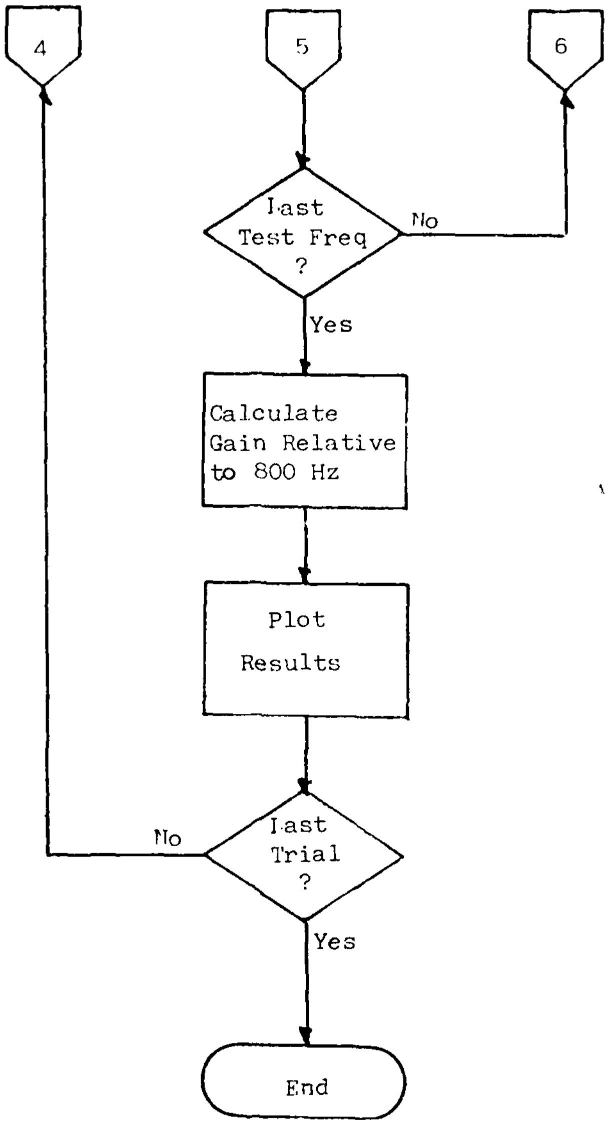

Figure 22. (continued) Program DGAIN Flowchart

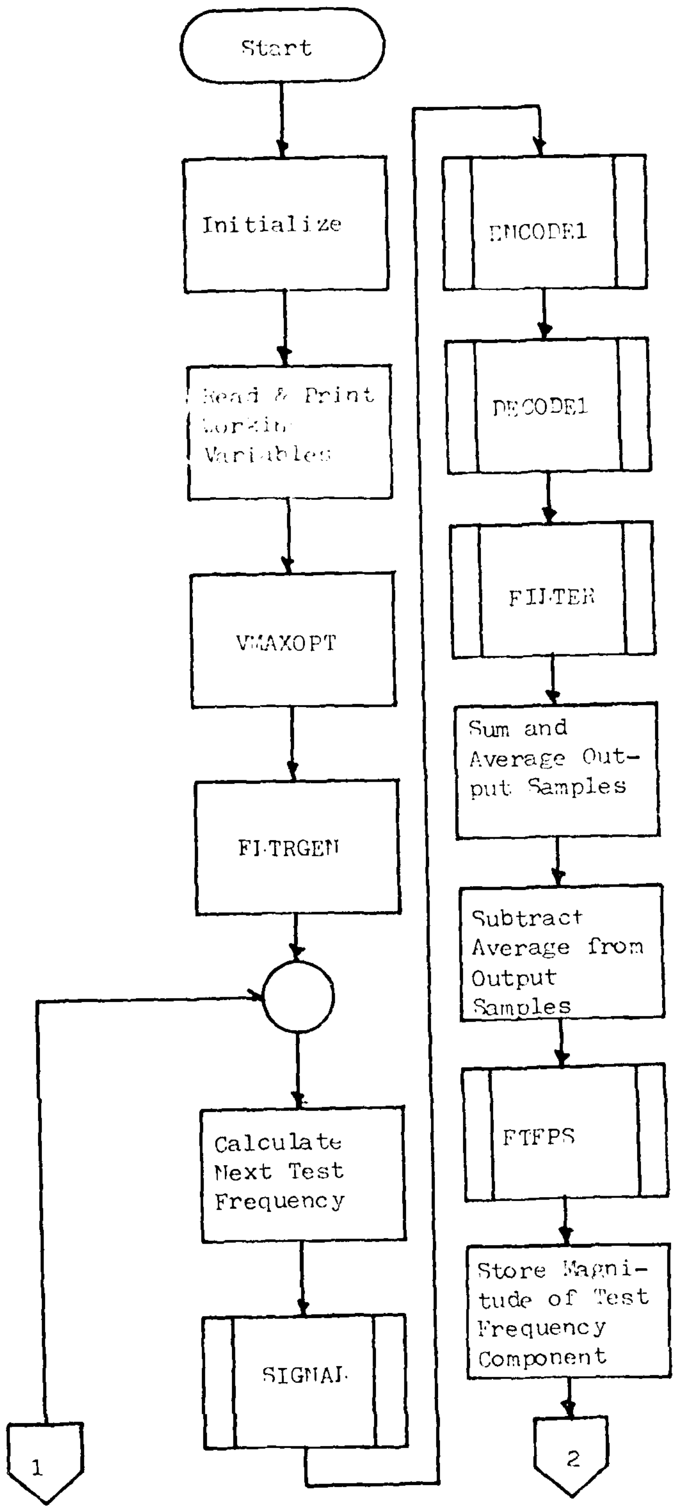

Figure 23. Flowchart of Frequency Selective Method of Frequency Response Measurement Program (RESP)

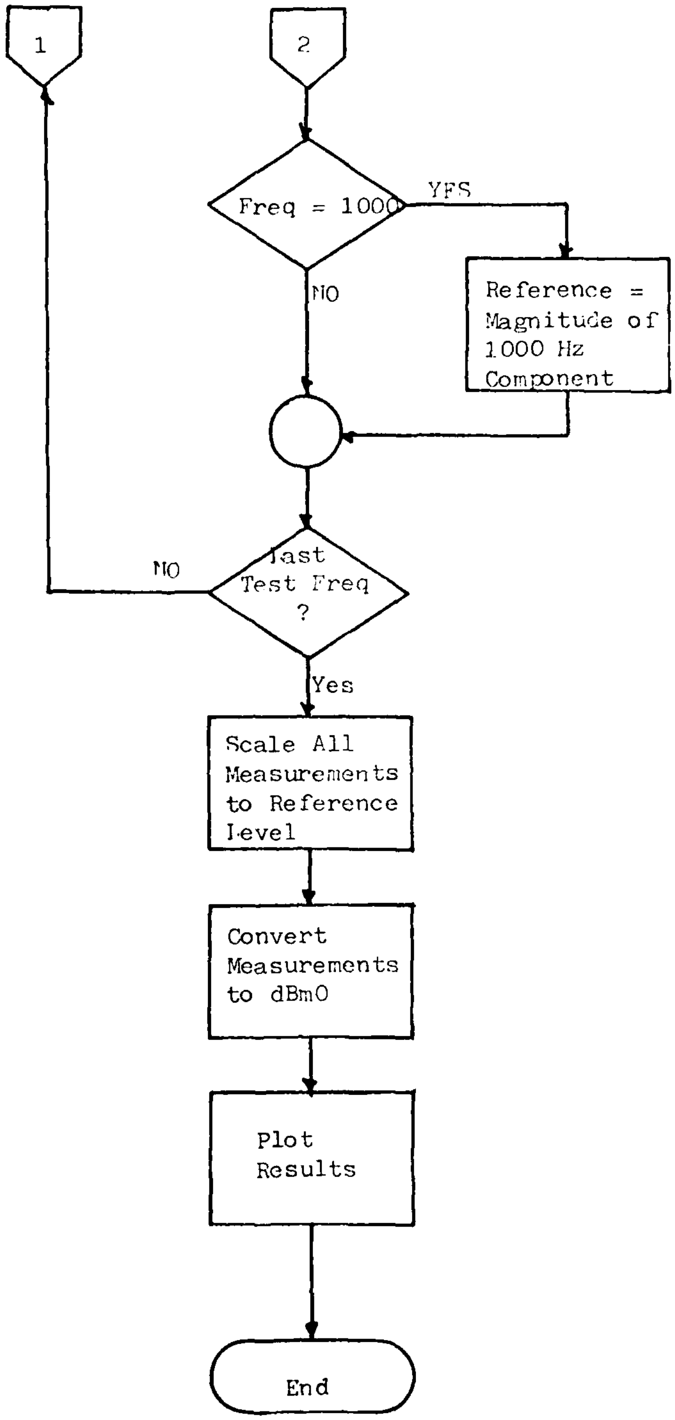

Figure 23. (continued) Program Flowchart (RESP)

reference value since the fast fourier transform used to calculate the output signal spectrum cannot measure the component at 800 Hz.

IV. Test Results

The results of the tests described in the previous section are presented here. Each test was first performed with the standard 800 Hz test signal, while the CVSD encoder and decoder parameters were matched. This test characterized the ideal system performance with the system parameters at their nominal values. Next, the tests were performed allowing the system parameters to vary across the ranges shown in Table I and using test signals that varied in frequency and power across their normal dynamic ranges, while still maintaining encoder/decoder match. Finally, the test were repeated again with the encoder parameters held constant at one extreme of the permissible values and the decoder parameters allowed to range to the opposite extreme. Each test was performed changing one variable at a time while the other were held at their nominal values.

Idle Channel Noise The results of the idle channel noise tests are shown in figure 24. For each sample rate, the idle channel noise performance improves as the step size ratio increases. This results from the decrease in minimum step size as the step size ratio increases. The output signal depends entirely on the minimum step size when the system input is zero or grounded. Since the minimum step size is defined to be non-zero, the output signal will alternate positive and negative around zero attempting to approximate the zero input signal. The smaller the deviation from zero, the less the power in the output signal and the better the idle channel noise performance. System performance exceeds the criteria specified in the draft standard. Idle channel noise is -88 dBm0 vs. the specified -50 dBm0 at 16 kb/s sample speed and -97 dBm0 vs. -60 dBm0 at 32 kb/s.

Encoder/decoder parameter mismatch has no effect on idle channel noise. This is a result of the fact that no matter what the encoder's parameters, the output will always be alternating ones and zeros when the encoder input is grounded. Therefore, the input signal at the decoder will always be the same and the output signal will only be affected by the decoder parameters. The idle channel noise performance under mismatched conditions will be the same as shown in figure 24 where the

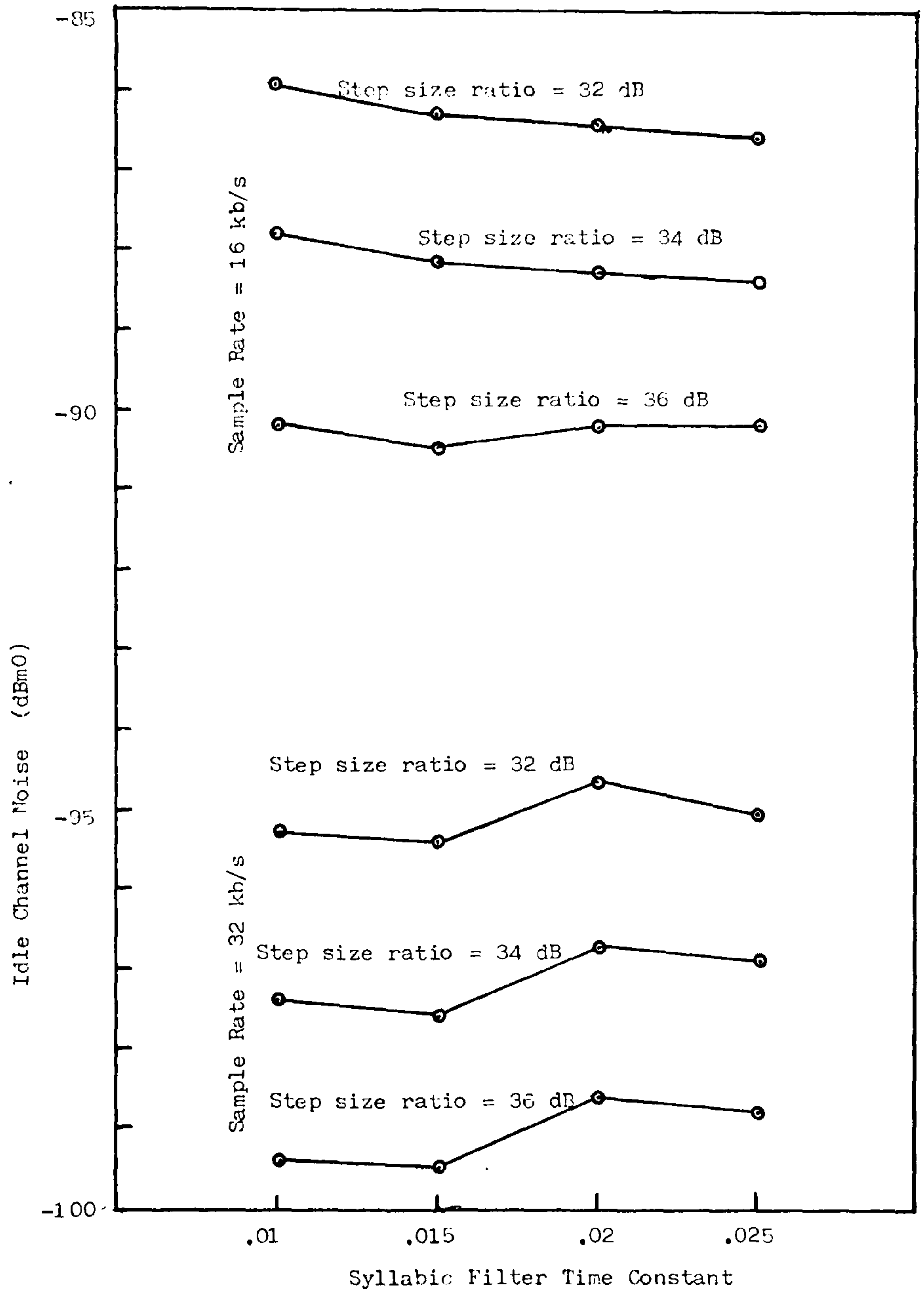

Figure 24. CVSD Signal Processing System Idle Channel Noise Performance

parameters are those of the decoder.

Total Harmonic Distortion The draft standard specifies no maximum total harmonic distortion for the CVSD system, however, it is generally accepted that distortion levels of less than 20% will not usually be objectionable to the system users. As can be seen from the test results shown in figure 25, system performance at the 16 kb/s sample rate exceeds this limit by 4-8%. System performance when the sample rate is increased to 32 kb/s improves substantially. The total harmonic distortion level drops to approximatley 6%. Figure 26 shows that the distortion level at both sample rates is relatively constant for all input power levels within the normal operating range except at the very low power levels. When the encoder and decoder parameters are mismatched, the total harmonic distortion performance shows some degradation as figures 27, 28, and 29 indicate. The largest amount of deviation from the matched system performance occurs at the very low power levels where the impact will have the least effect. As figure 29 shows, total harmonic distortion is most sensitive to mismatches of the encoder and decoder primary integrator pole frequencies. Syllabic filter time constants and step size ratios have minimal impact on the system performance when mismatched, however, all have the most impact at the very low input power levels.

Intermodulation Distortion Intermodulation distortion performance for the system model with nominal parameter values is shown in figure 30. As is the case with the total harmonic distortion test, the draft standard provides no perfromance criteria. In general, intermodulation levels of more than 4-5% will be objectionable to a system user. At the 16 kb/s sample rate, the intermodulation distortion measured ranges from 1 to 5% depending on the syllabic filter time constant used. The distortion falls to approximately 1% when the sample rate is increased to 32 kb/s. Figure 31 shows the system intermodulation response as the input signal power is varied. System non-linearities cause the distortion levels to rise at the very low signal levels and at the high input power levels. Across the normal operating levels between -10 dBm0 and -30 dBm0, the distortion is generally less than 5%. When the encoder and

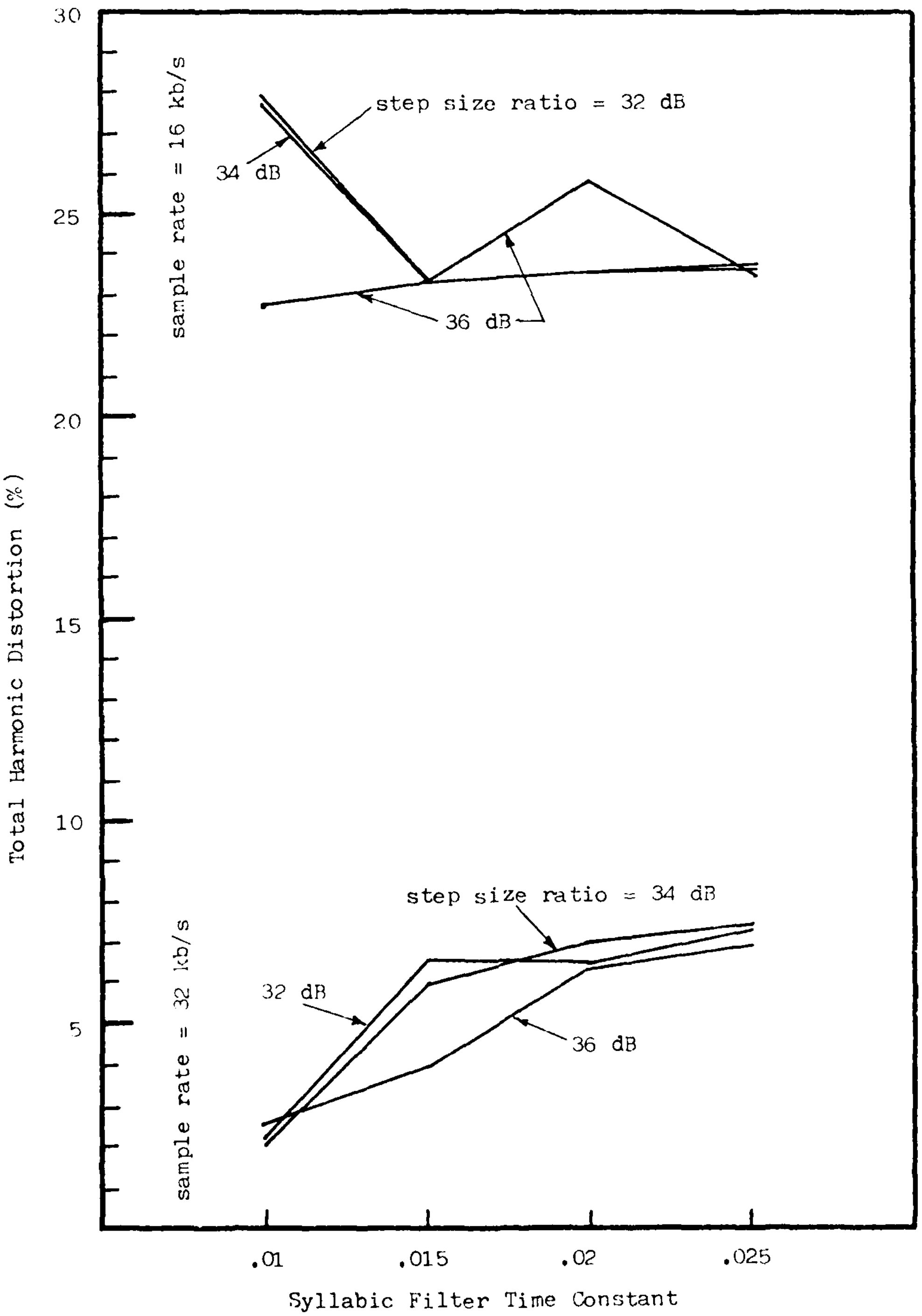

Figure 25. CVSD Signal Processing System Total Harmonic Distortion Performance with Encoder and Decoder Parameters Matched (Test Signal = 1000 Hz, -20dBm0)

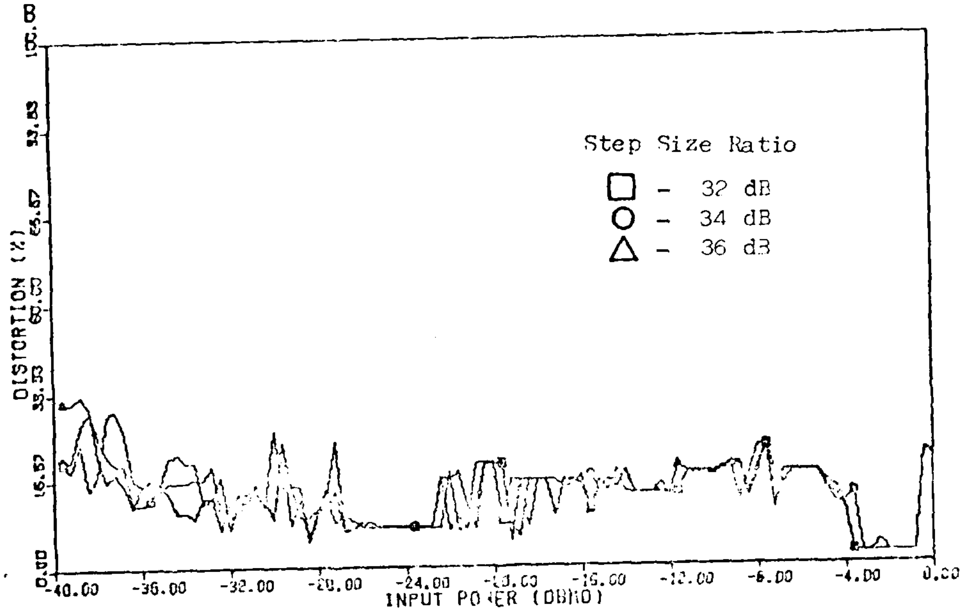

Figure 26a. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Parameters Matched at 16 kb/s Sample Rate (1000 Hz Test Signal)

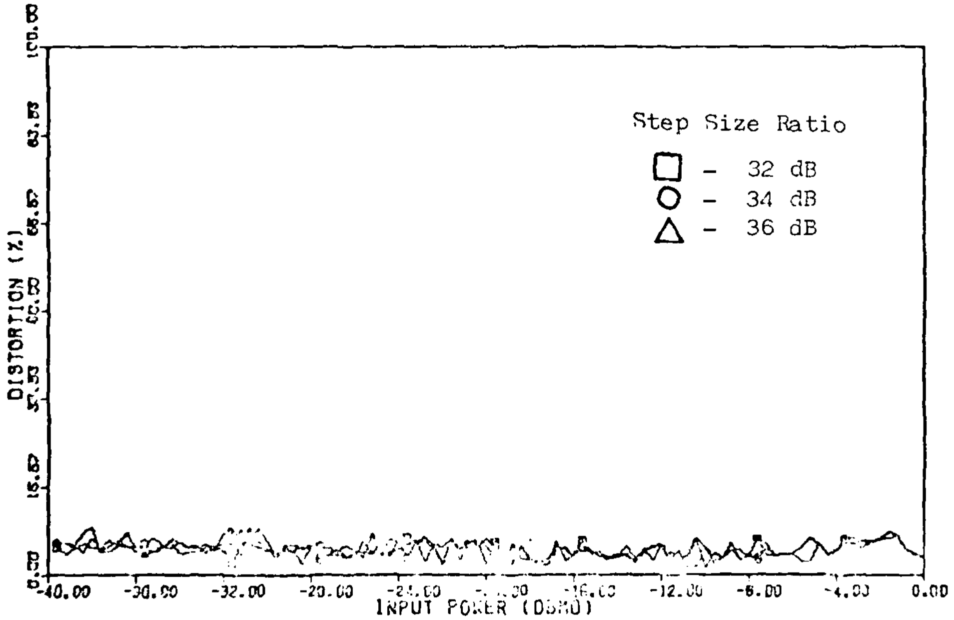

Figure 26b. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Parameters Matched at 32 kb/s Sample Rate (1000 Hz Test Signal)

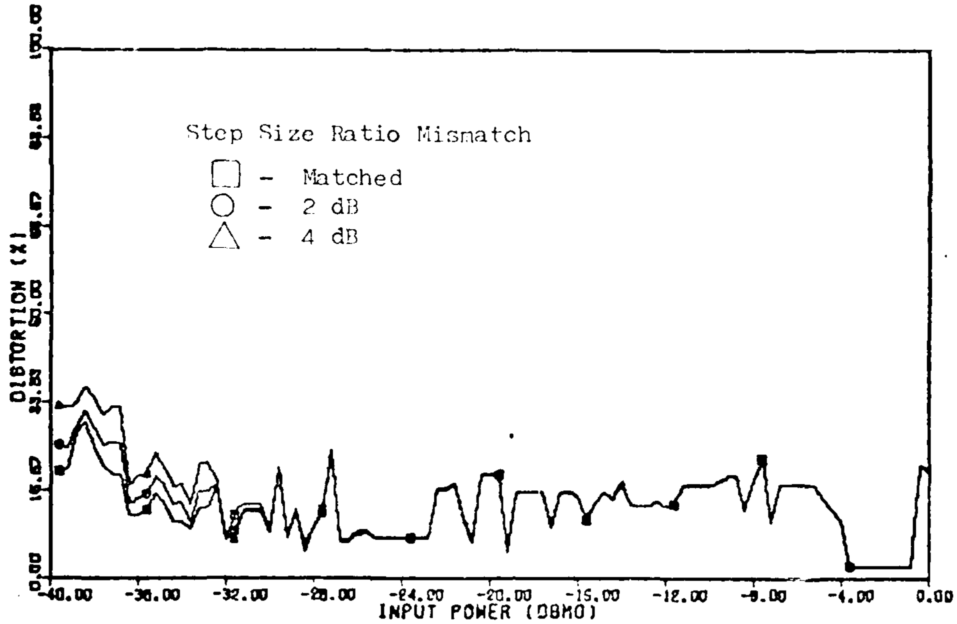

Figure 27 a. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Step Size Ratios Mismatched at 16 kb/s Sample Rate (1000 Hz Test Signal)

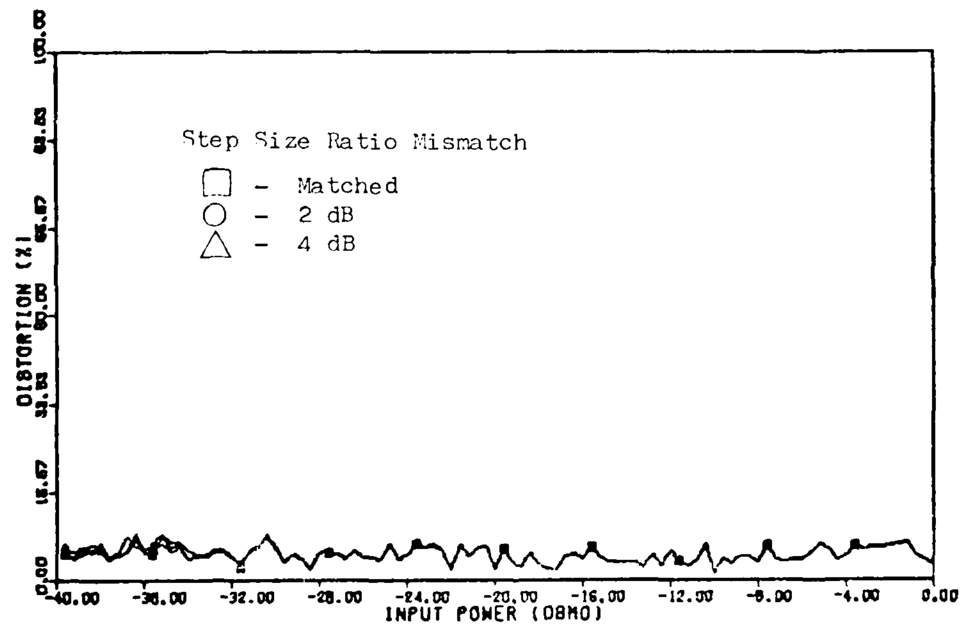

Figure 27 b. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Step Size Ratios Mismatched at 32 kb/s Sample Rate (1000 Hz Test Signal)

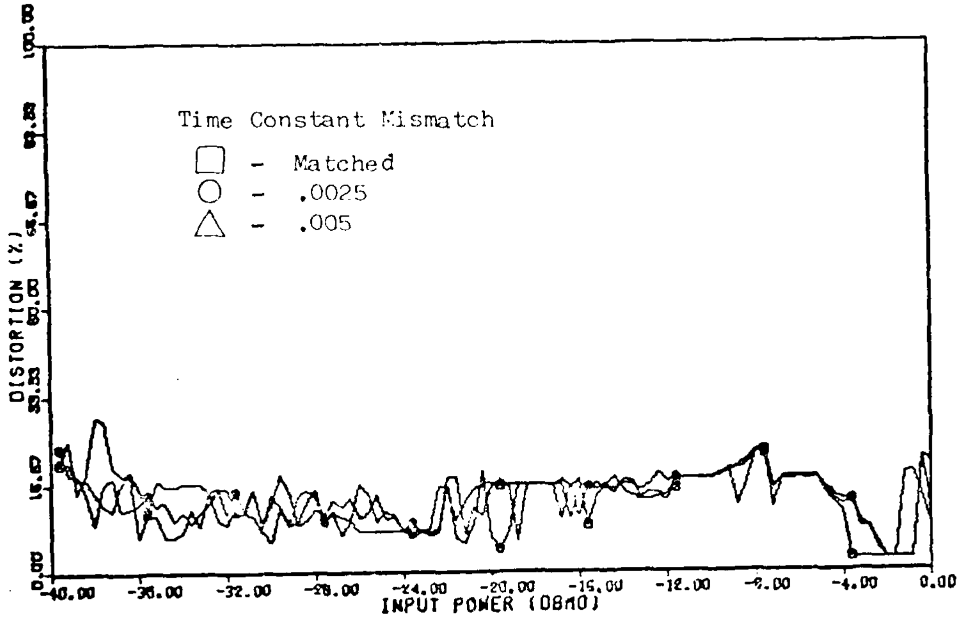

Figure 28 a. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (1000 Hz Test Signal)

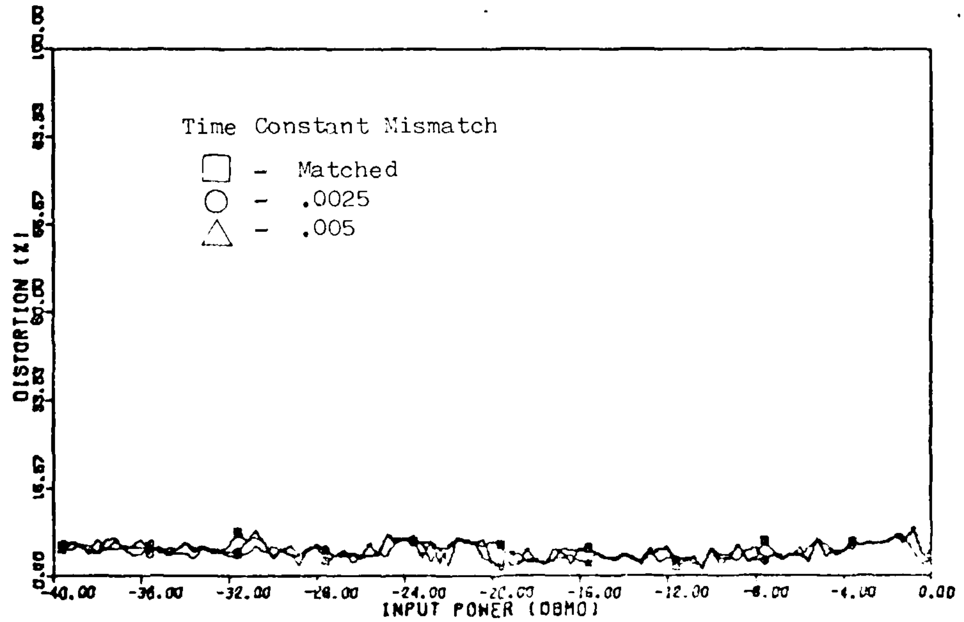

Figure 28 b. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (1000 Hz Test Signal)

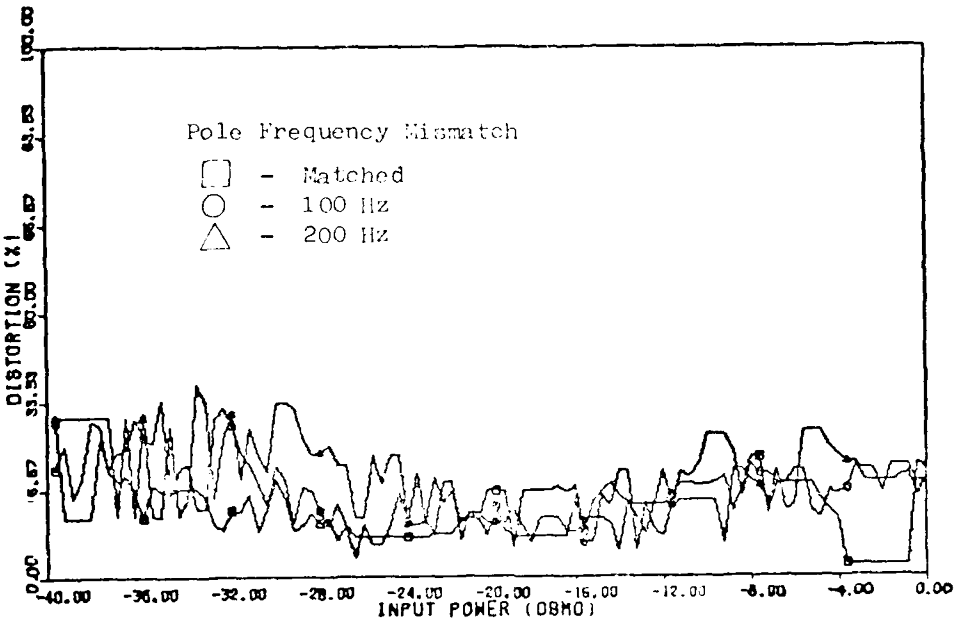

Figure 29 a. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (1000 Hz Test Signal)

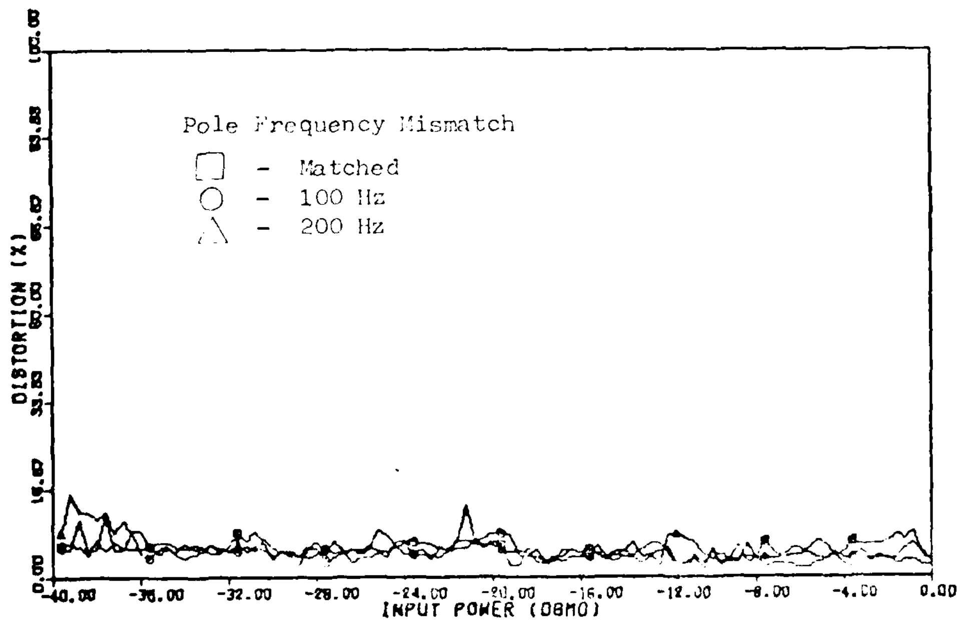

Figure 29 b. CVSD System Total Harmonic Distortion Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (1000 Hz Test Signal)

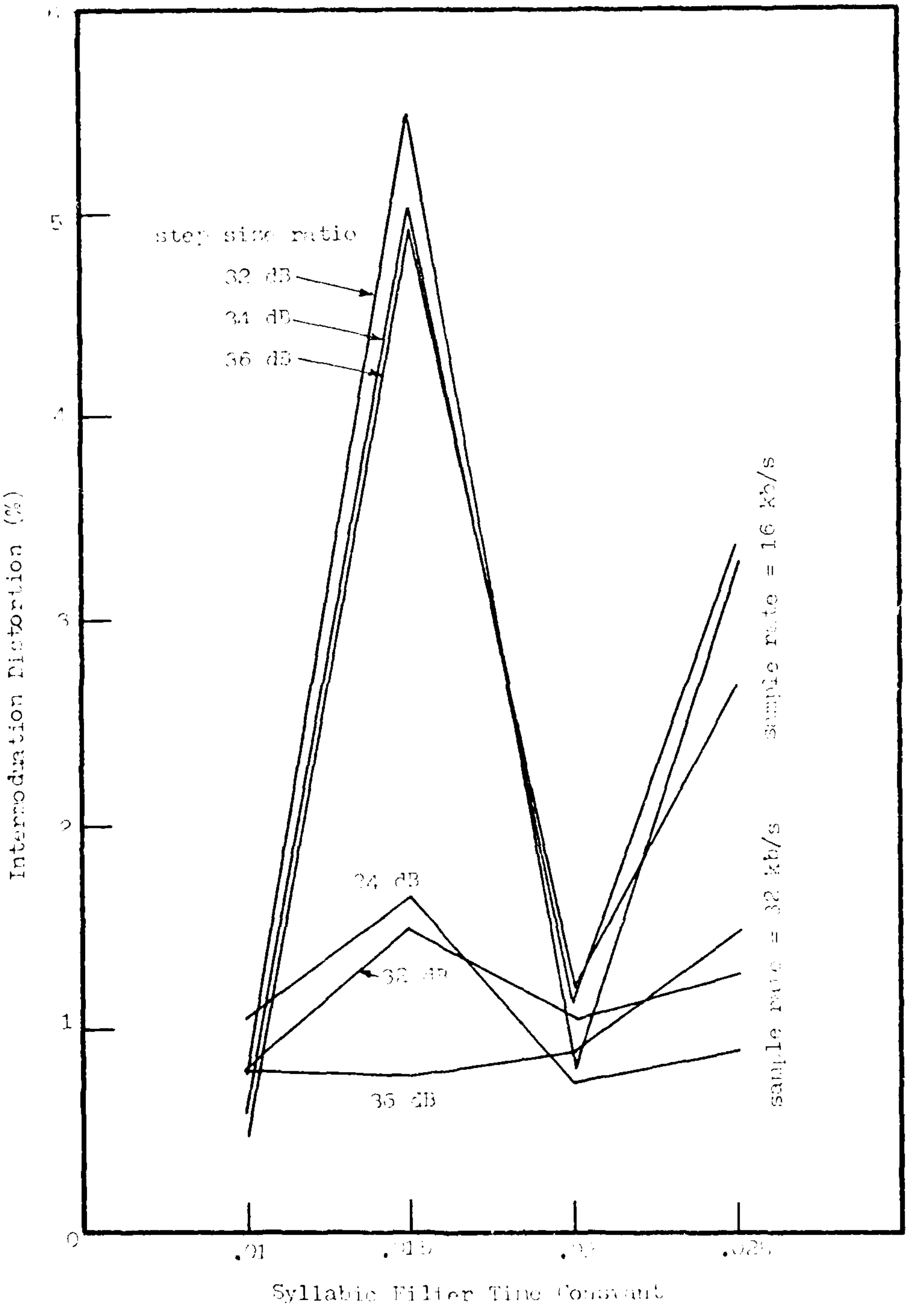

Figure 30. CVSD Signal Processing System Intermodulation Distortion Performance with Encoder and Decoder Parameters Matched (Test Signal = 1000 Hz, -23 dBm0 and 750 Hz, -23 dBm0)

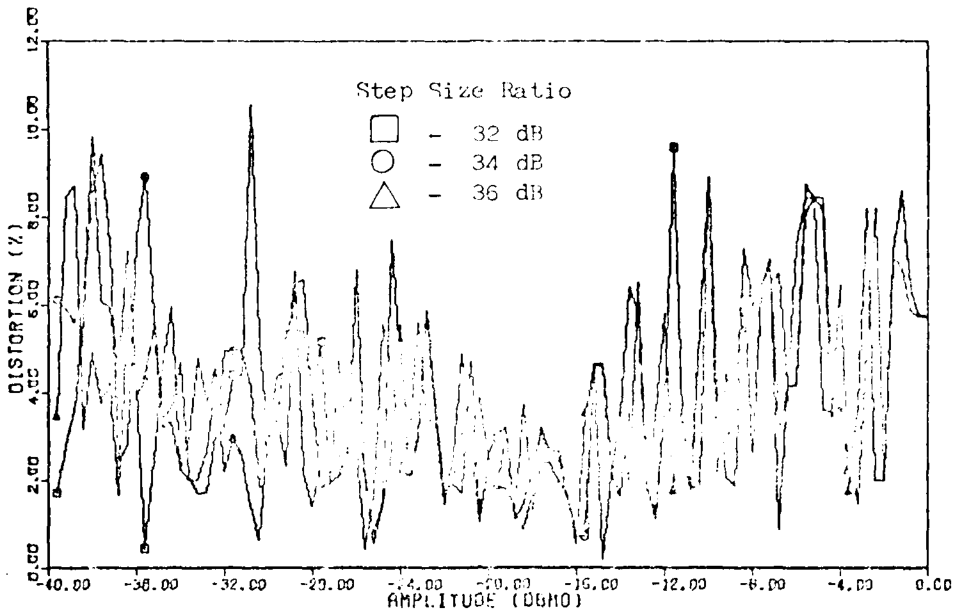

Figure 31a. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Parameters Matched at 16 kb/s (750 and 1000 Hz Test Signal)

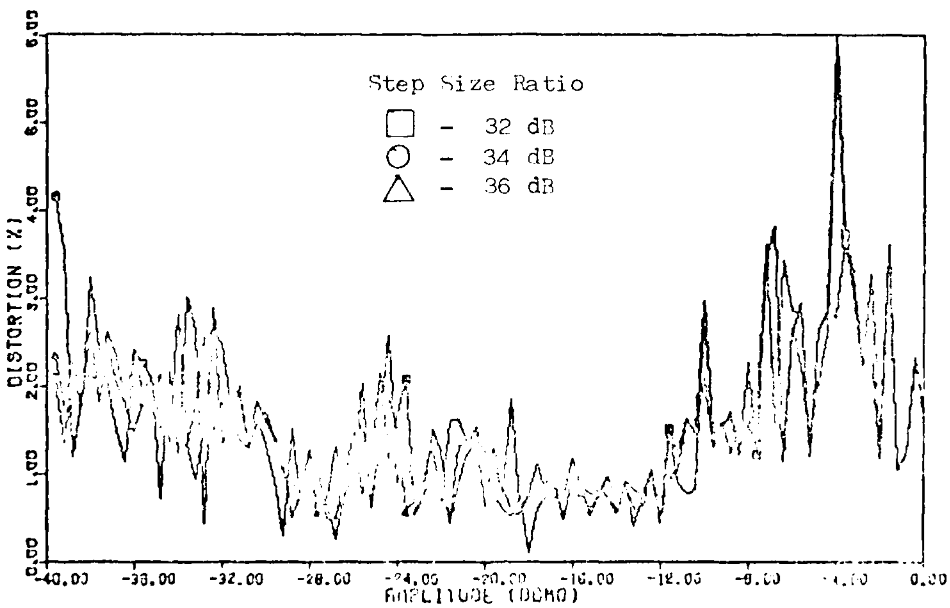

Figure 31b. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Parameters Matched at 32 kb/s Sample Rate (750 and 1000 Hz Test Signal)

and decoder are mismatched, the intermodulation distortion performance shows the same characteristics as the total harmonic distortion. The step size ratio and syllabic filter time constant have a minimal impact on system performance as shown by the results in figures 32 and 33. When the primary integrators have differing pole frequencies, the intermodulation distortion measurements show more deviation from the matched system performance as shown in figure 34.

<u>Signal-to-Noise Ratio</u> Figures 35 to 43 show how the system model signal-to-noise performance changes with variations is system parameters and input test signals. The SNR performance under matched conditions with the standard test signal shows very little variation with differing values of step size ratio, syllabic filter time constant, and primary integrator pole frequency. Signal-to-noise ratio vs. input frequency performance meets the criteria set by the draft stnadard across most of the voice band. Encoder/decoder mismatches of step size ratio and syllabic filter time constant have very little impact on system performance. A mismatch of the primary integrator pole frequencies, however, have a much larger effect on system performance. The SNR is degraded below the criteria set by the draft standard, with the largest deviation from the ideal performance occuring at the lower frequencies. At the 16 kb/s sample rate the SNR is lowered by about 5 dB and at the 32 kb/s speed, about 4 dB.

Signal-to-noise ratio performance vs. input signal power fails to meet the criteria established in the draft standard. At input levels below approximately -10 dBm0 the model's performance falls below the desired level. Trends in system performance as the result of variations in the system parameters can be observed in spite of this poor performance, as shown in figures 4[illegible] to [illegible]. Matched system performance minimal change as the system parameters are varied across the ranges specified in Table I. When the encoder and decoder are not matched, SNR, like total harmonic distortion and intermodulation distortion, shows little deviation from the matched system performance except at the very low signal levels. SNR shows the most change from ideal system performance when the primary integrator pole frequencies are mismatched.

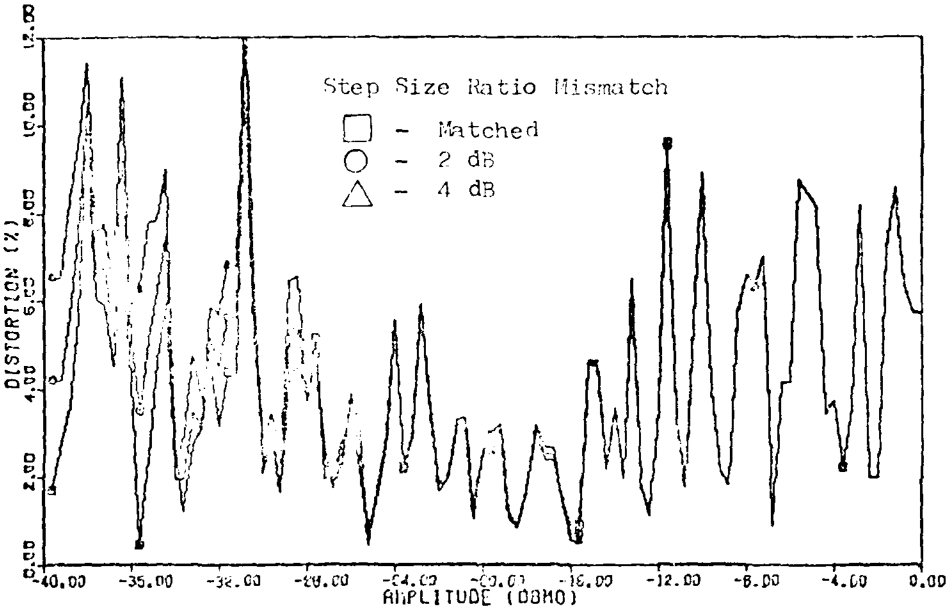

Figure 32a. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder And Decoder Step Size Ratios Mismatched at 16 kb/s Sample Rate (750 and 1000 Hz Test Signal)

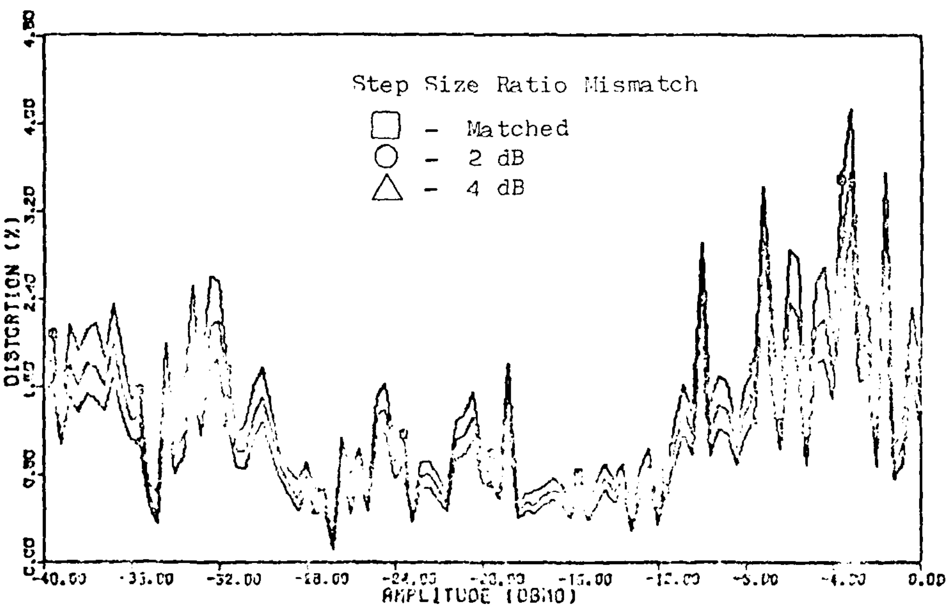

Figure 32b. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Step Size Ratios Mismatched at 32 kb/s Sample Rate (750 and 1000 Hz Test Signal)

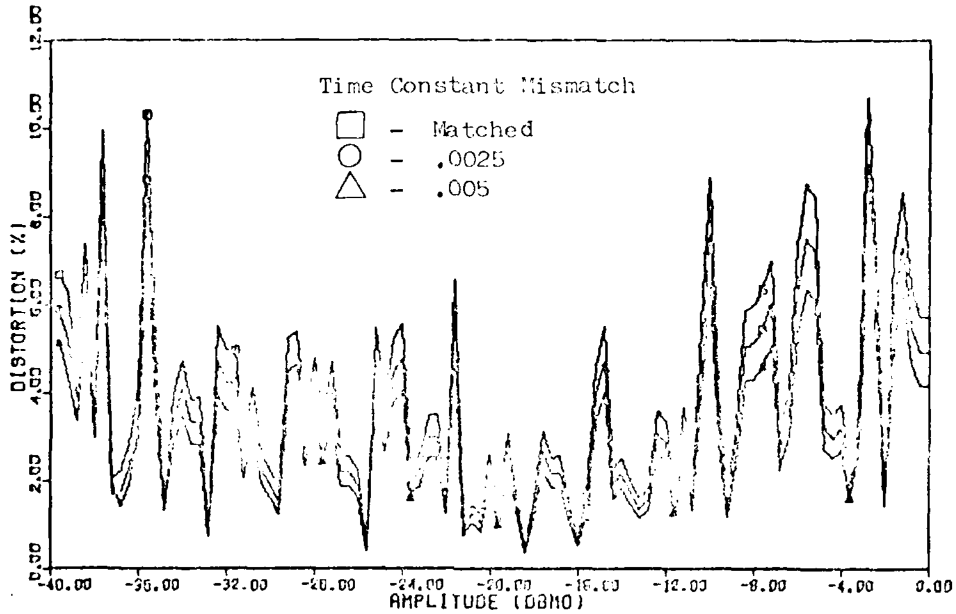

Figure 33a. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (750 and 1000 Hz Test Signal)

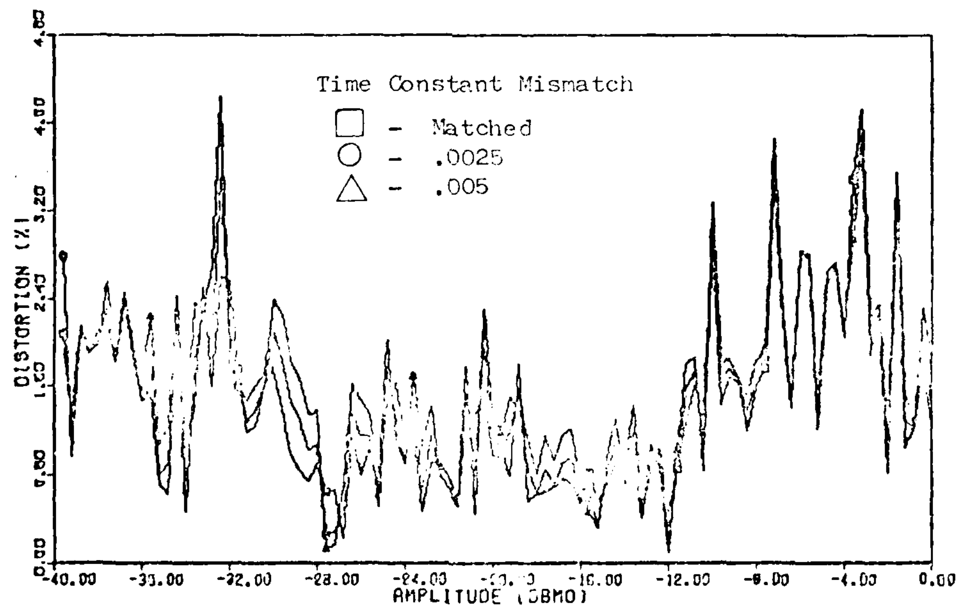

Figure 33b. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (750 and 1000 Hz Test Signal)

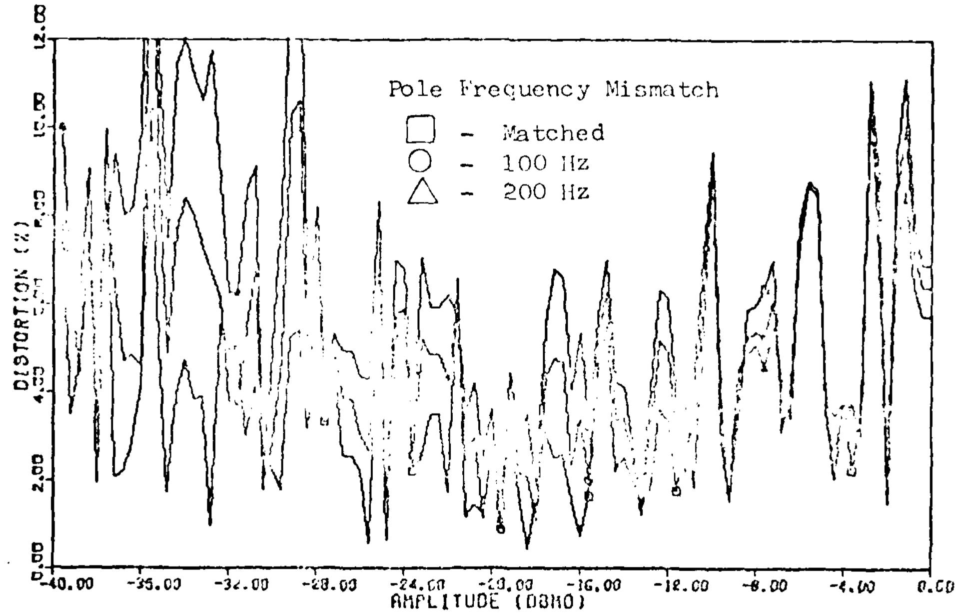

Figure 34a. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (750 and 1000 Hz Test Signal)

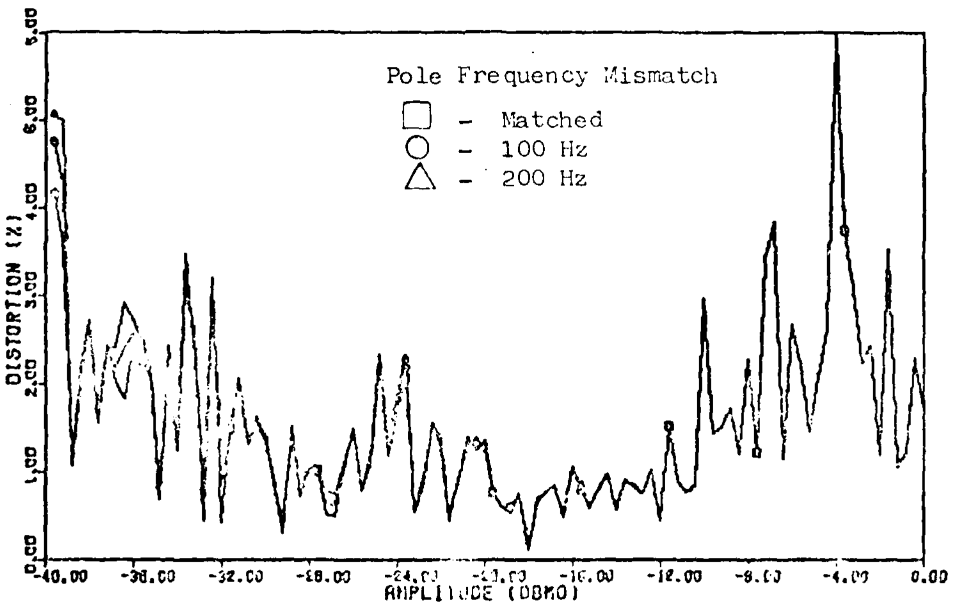

Figure 34b. CVSD System Intermodulation Distortion Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (750 and 1000 Hz Test Signal)

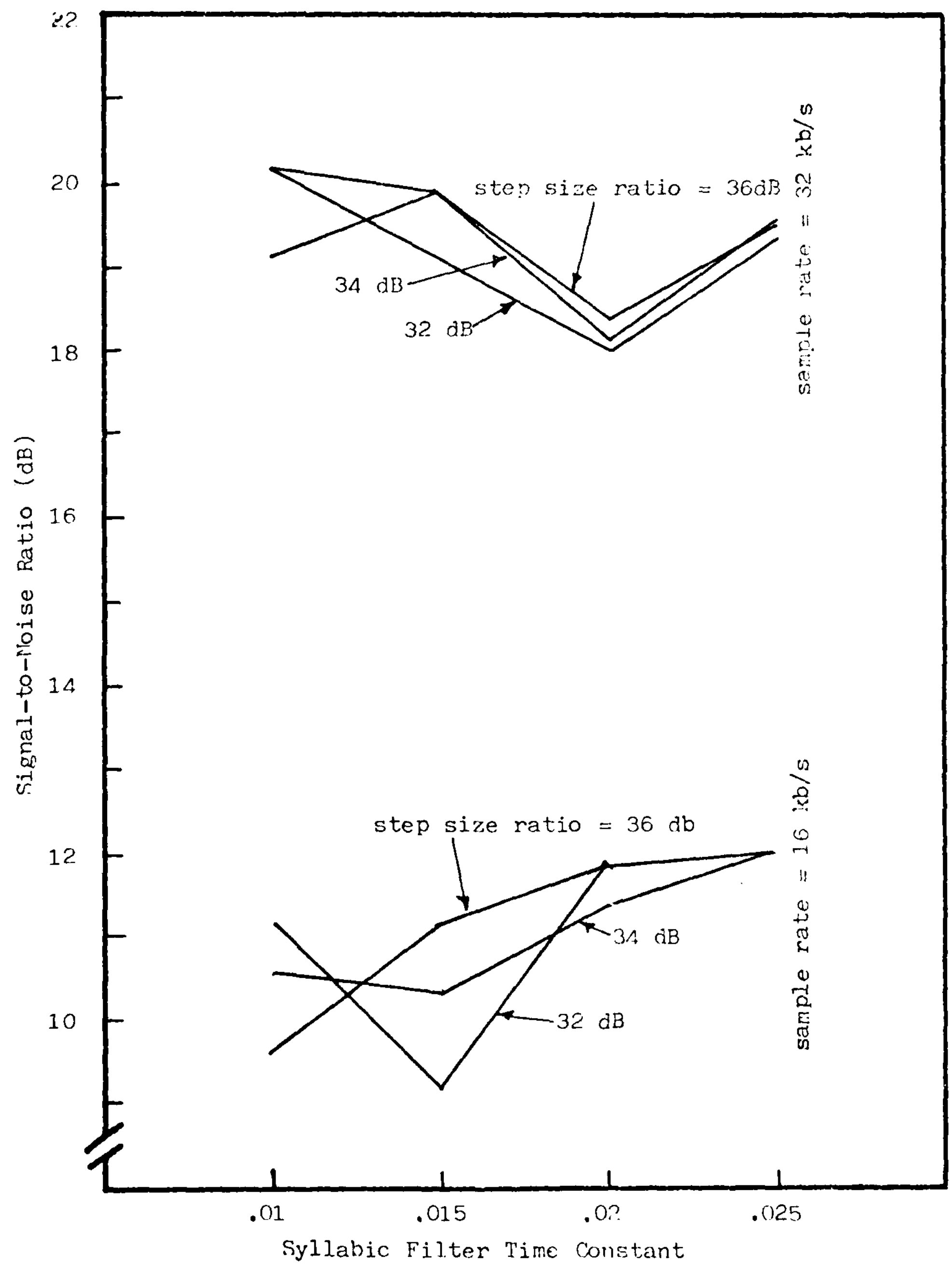

Figure 35. CVSD Signal Processing System Signal-to-Noise Performance with Encoder and Decoder Parameters Matched (Test Signal = 800 Hz, -20dBm0)

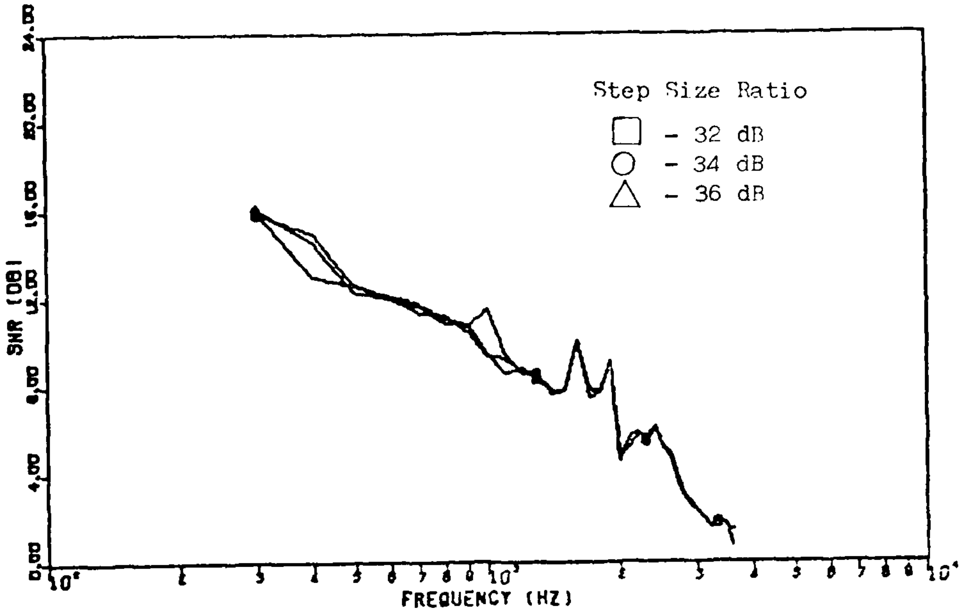

Figure 36 a. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Parameters Matched (-20 dBm0 test signal) at 16 kb/s Sample Rate

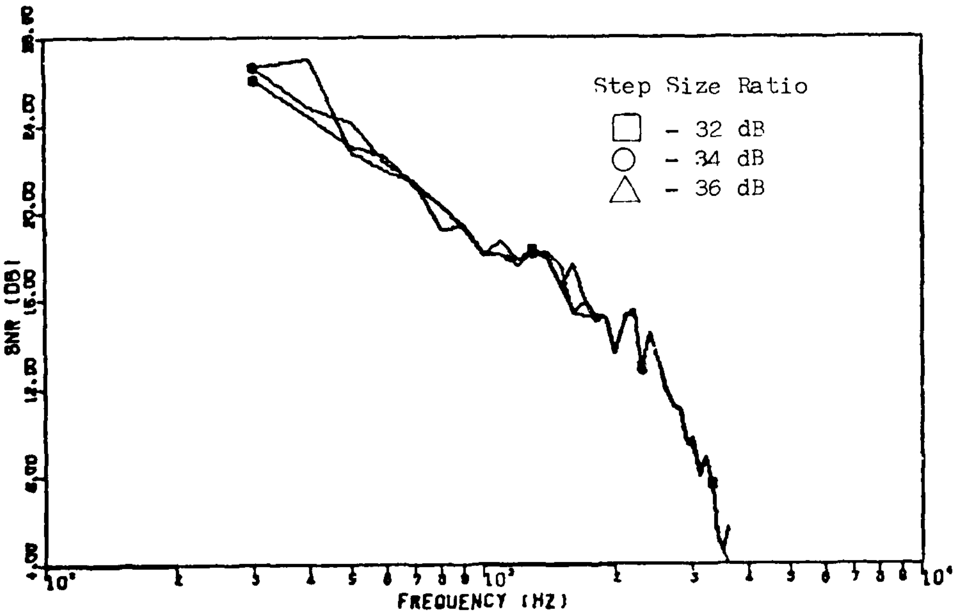

Figure 36b. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Parameters Matched (-20 dBm0 test signal) at 32 kb/s Sample Rate

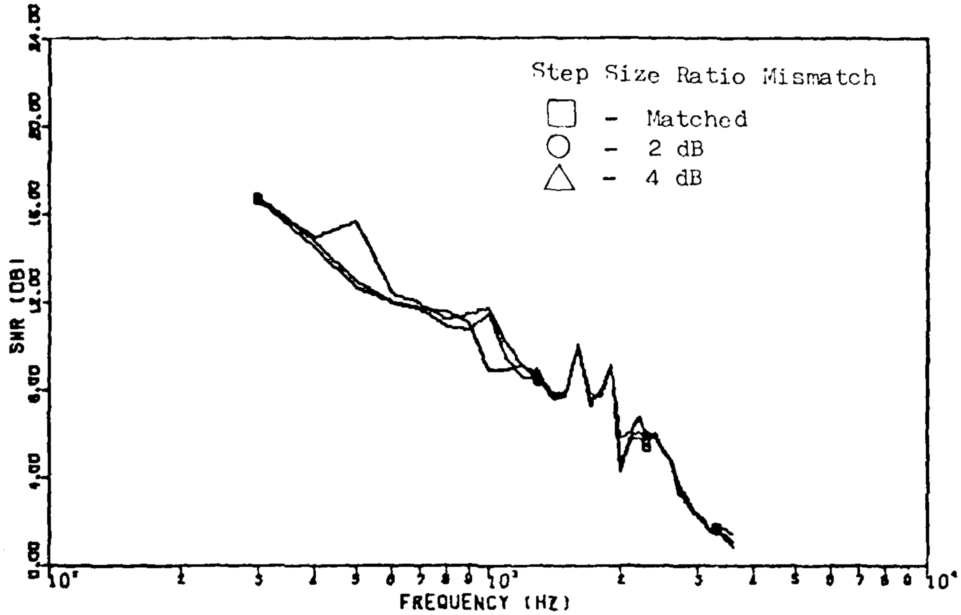

Figure 37a. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Step Size Ratio Mismatched at 16 kb/s Sample Rate (-20 dBm0 test signal)

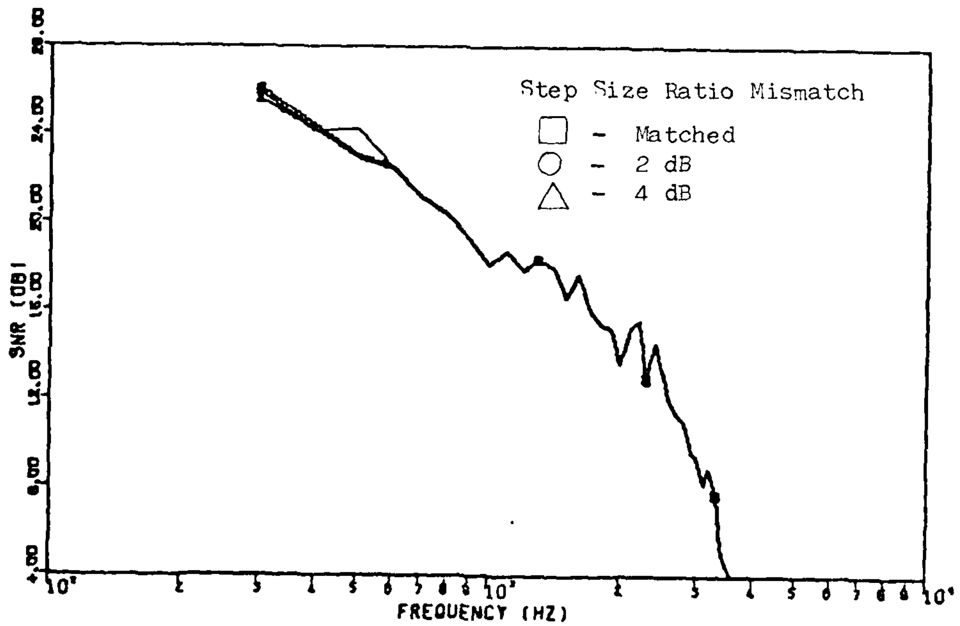

Figure 37b. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Step Size Ratio Mismatched at 32 kb/s Sample Rate (-20 dBm0 test signal)

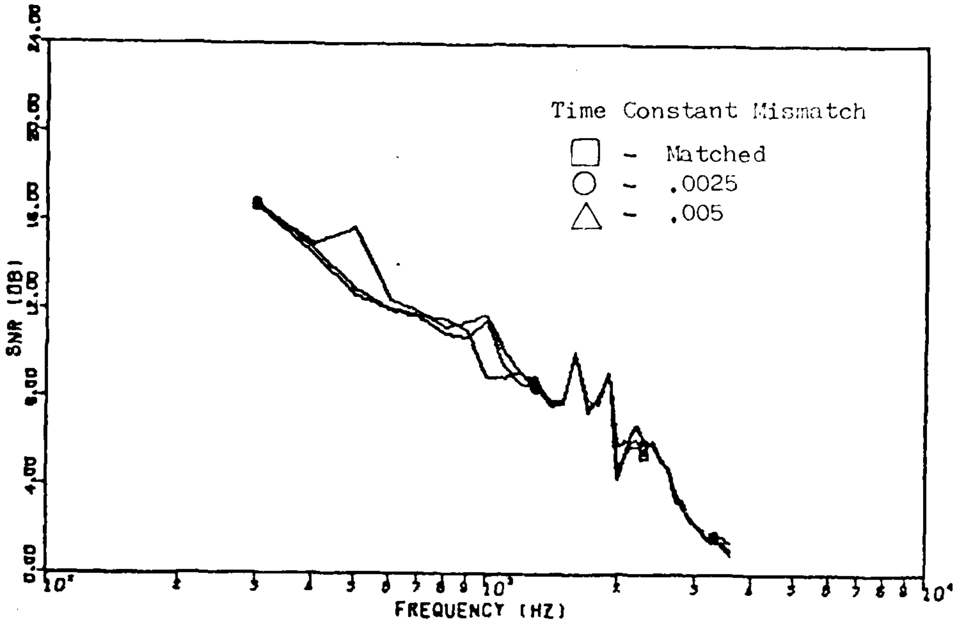

Figure 38 a. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

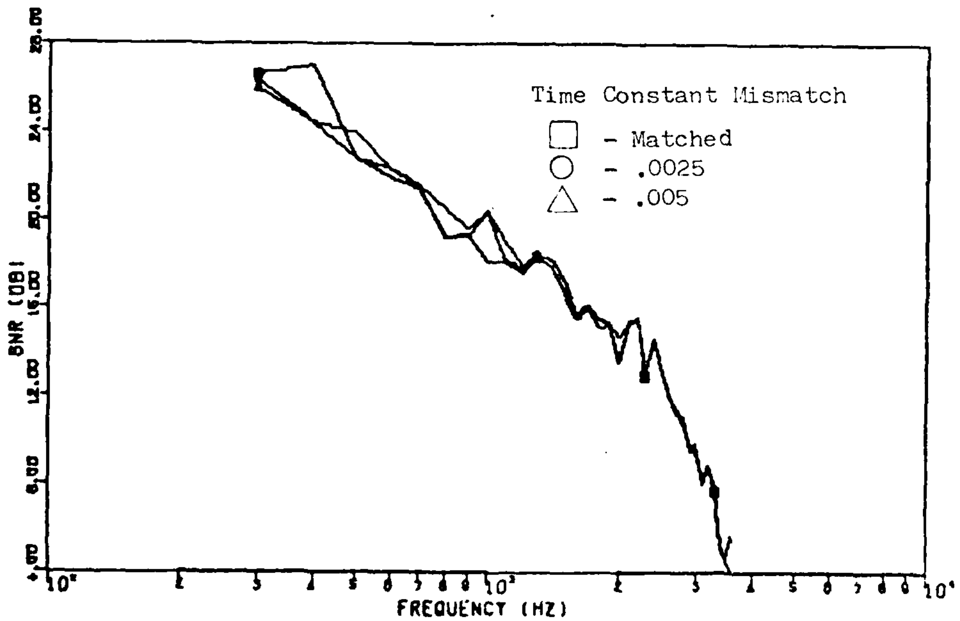

Figure 38 b. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

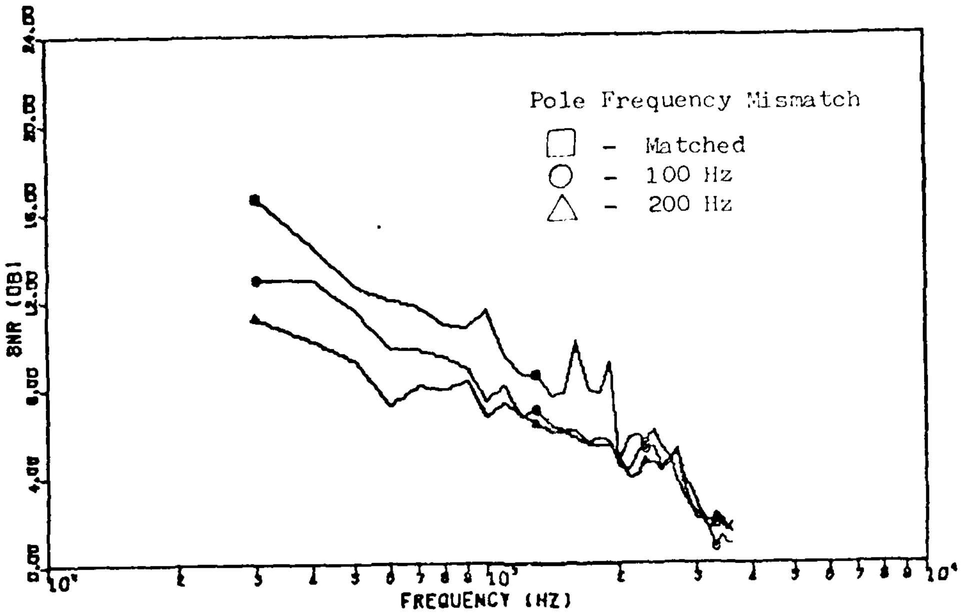

Figure 39 a. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

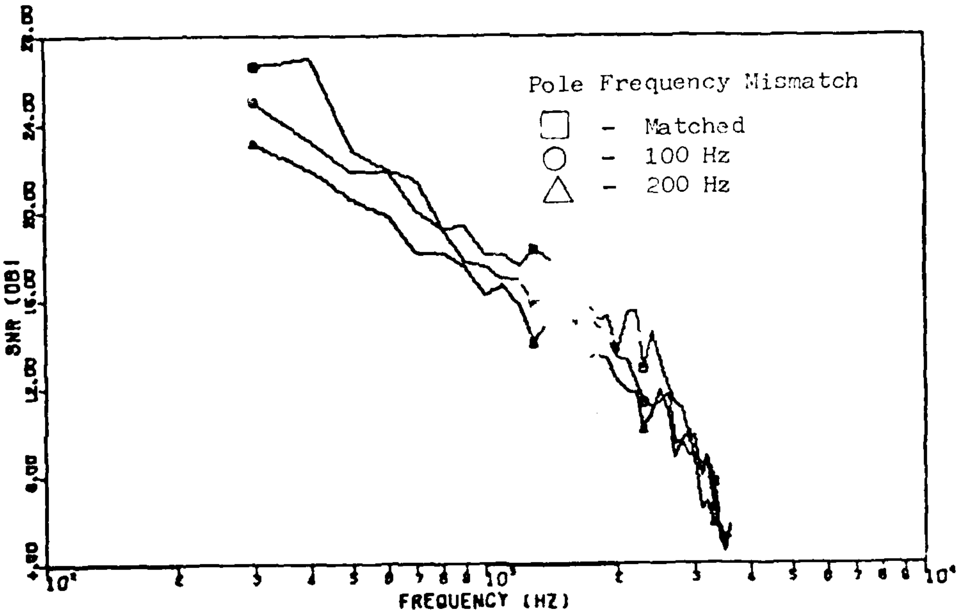

Figure 39 b. CVSD System Signal-to-Noise Performance vs. Frequency with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

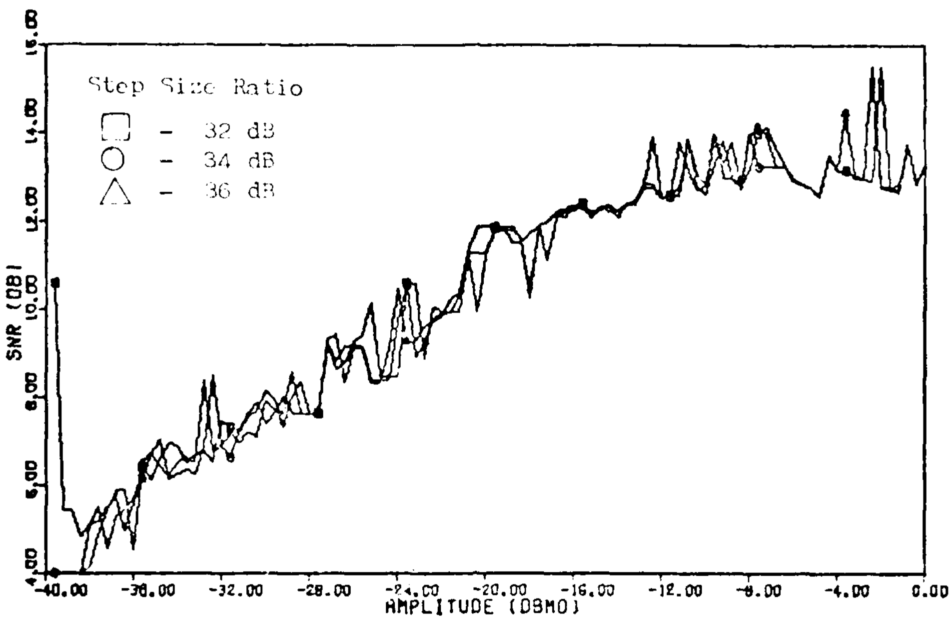

Figure 40a. CVSD System Signal-to-Noise Performance vs. Input Signal Amplitude with Encoder and Decoder Parameters Matched at 16 kb/s Sample Rate (800 Hz test signal)

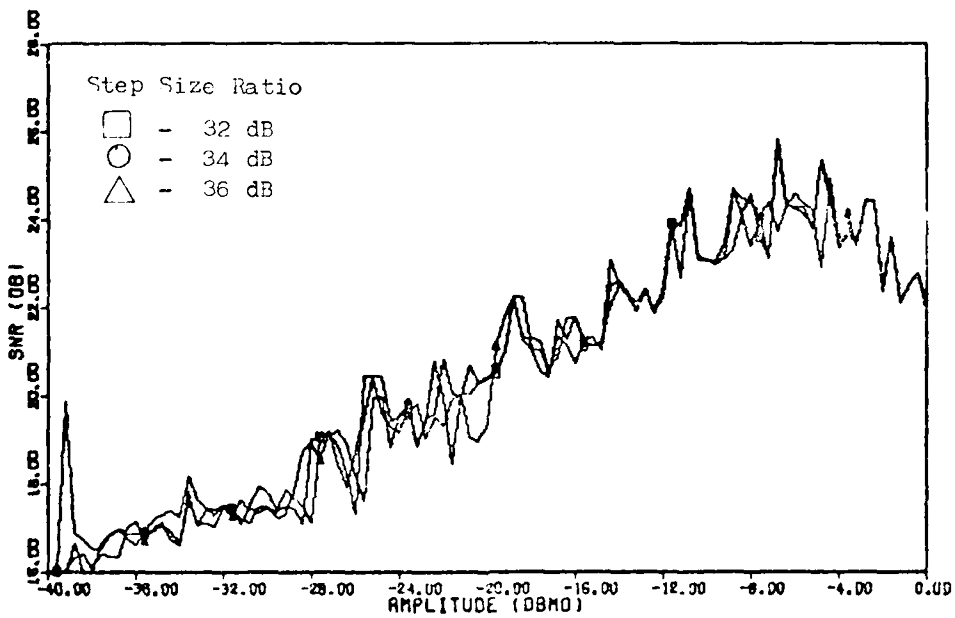

Figure 40b. CVSD System Signal-to-Noise Performance vs. Input Signal Amplitude with Encoder and Decoder Parameters Matched at 32 kb/s Sample Rate (800 Hz test signal)

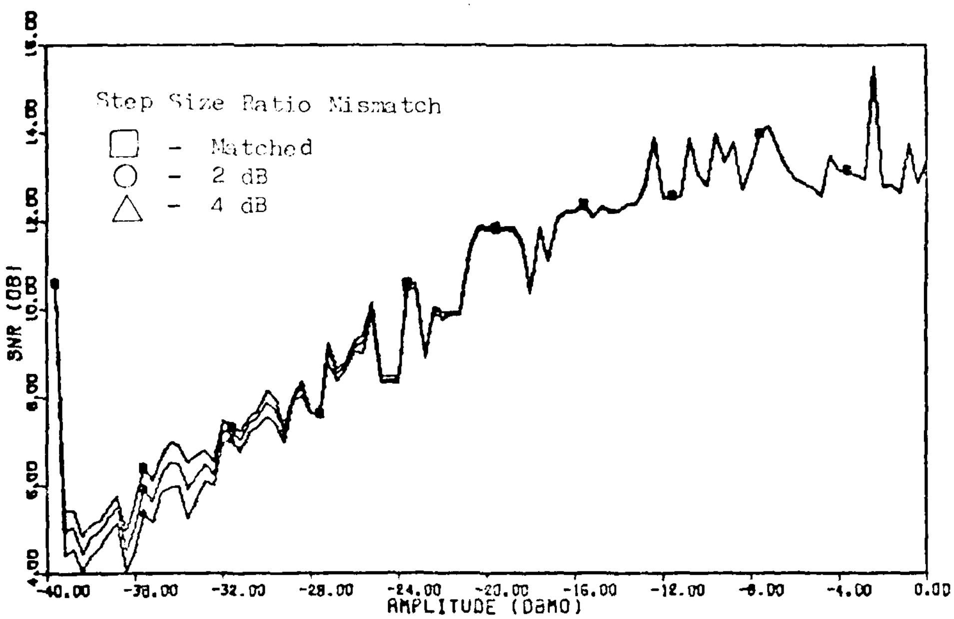

Figure 41a. CVSD System Signal-to-Noise Performance vs. Input Signal Amplitude with Encoder and Decoder Step Size Ratio Mismatched at 16 kb/s Sample Rate (800 Hz test signal)

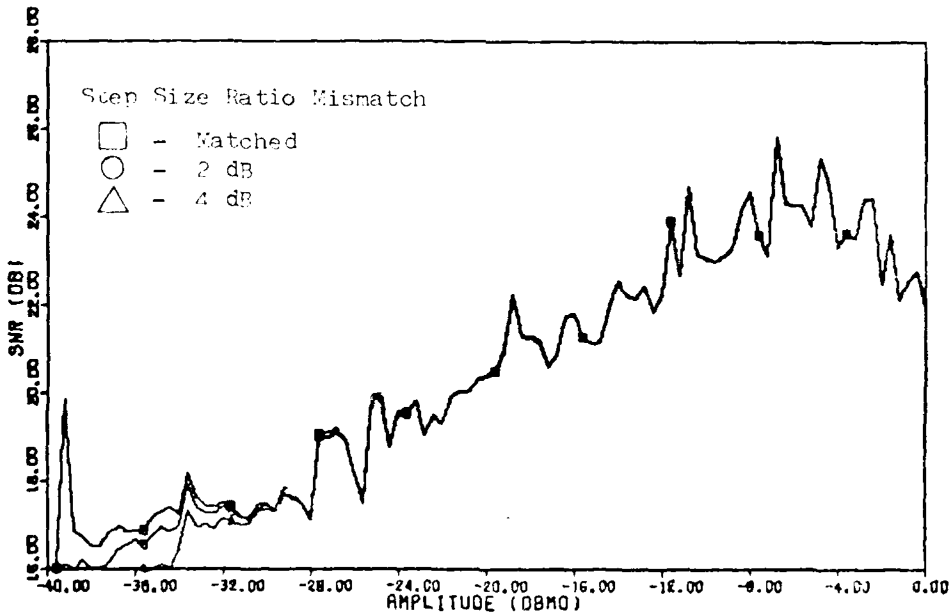

Figure 41b. CVSD System Signal-to-Noise Performance vs. Input Signal Amplitude with Encoder and Decoder Step Size Ratio Mismatched at 32 kb/s Sample Rate (800 Hz test signal)

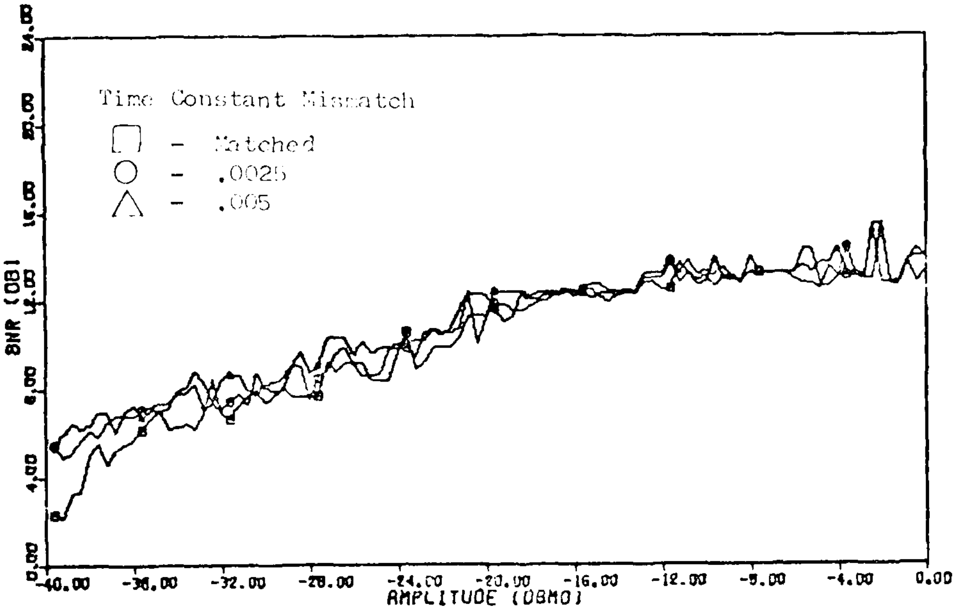

Figure 42a. CVSD System Signal-to-Noise Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (800 Hz Test Signal)

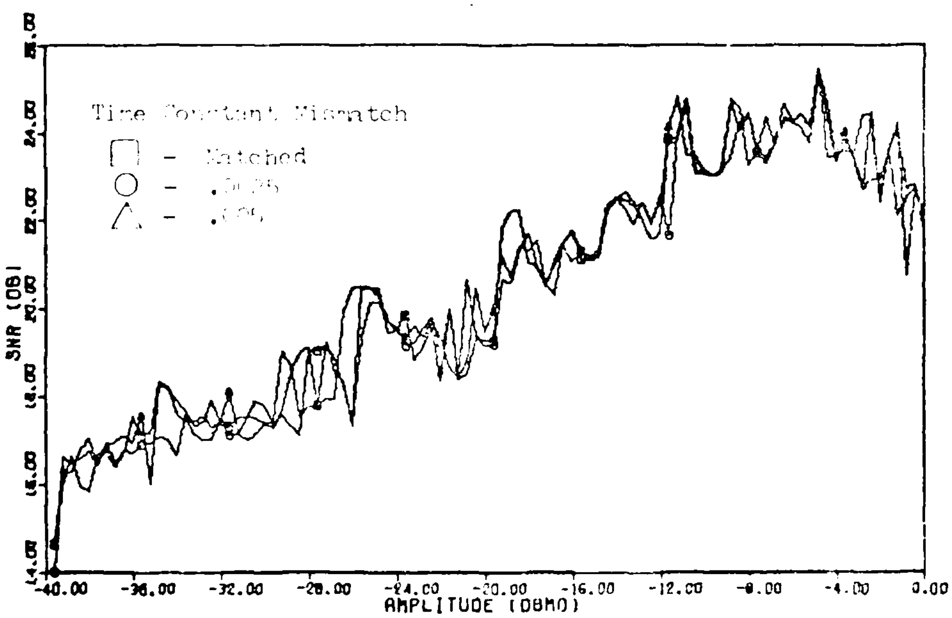

Figure 42b. CVSD System Signal-to-Noise Performance vs. Input Signal Power with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (800 Hz Test Signal)

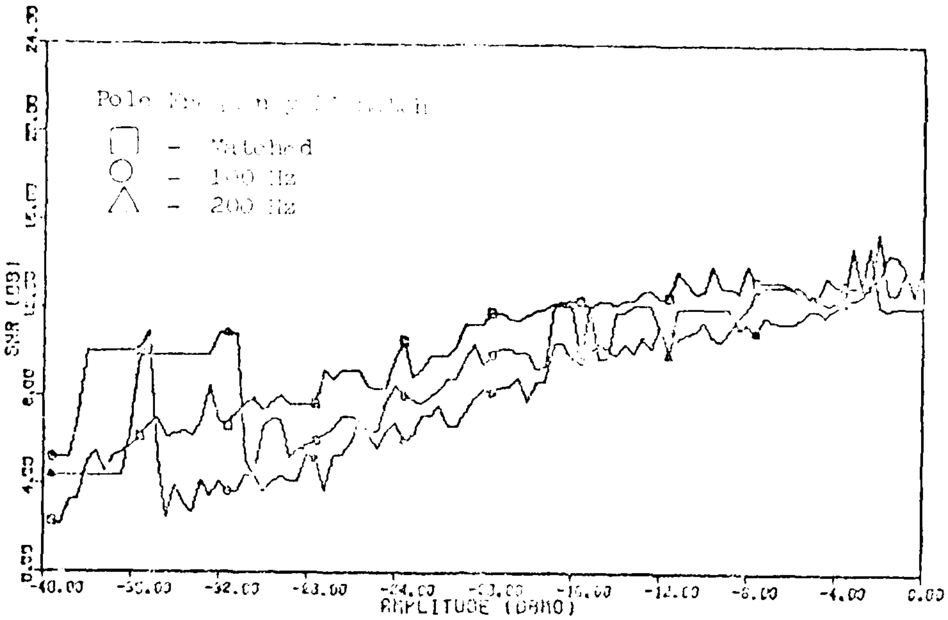

Figure 43a. CVSD System Signal-to-Noise Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (800 Hz Test Signal)

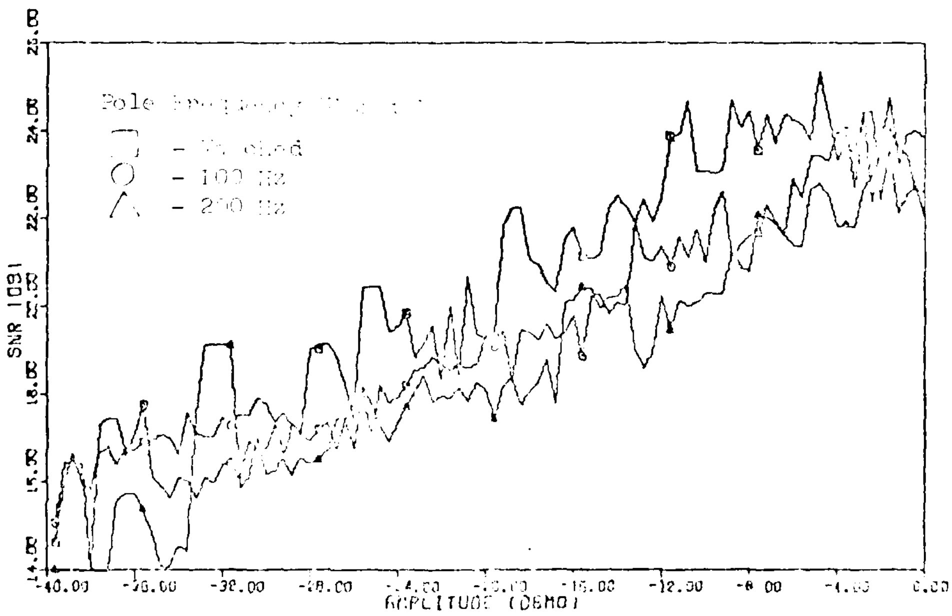

Figure 43b. CVSD System Signal-to-Noise Performance vs. Input Signal Power with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (800 Hz Test Signal)

Frequency Response The system frequency response characteristics are specified in the draft standard when measured by the flat weighted method. The results of the tests performed using this technique are shown in figures 44 to 47. The computer model's performance complies with the specified criteria except at the top end of the voice band. The model's response rolls off sharply at approximately 3 kHz. While the draft standard allows some roll-off, the response is not allowed to break sharply until 6 kHz. Variations in the system parameters have very little effect on the response characteristics when the encoder and decoder are matched. Under mismatched conditions, frequency response performance follows the same pattern as that established in the previously described tests. The step size ratio and syllabic filter time constant have minimal impact, while the primary integrator pole frequency causes increased deviation from the matched system performance.

Using the frequency selective measurement technique, the response characteristics of the CVSD encoder and decoder connected back-to-back without the input or output filters were measured. The results are shown in figures 48 to 51. These tests show that the response rolls off at ¼ the sample frequency at both the 16 and 32 kb/s sample rates. At 4 kHz for the 16 kb/s sample rate and 8 kHz for the 32 kb/s sample rate the frequency response curves break sharply. Encoder and decoder mismatch has practically no effect on the frequency response of these system components. The primary integrator pole frequency shows slightly more effect on the response than the other parameters. Its effect is mainly at the very low frequencies in the voice band.

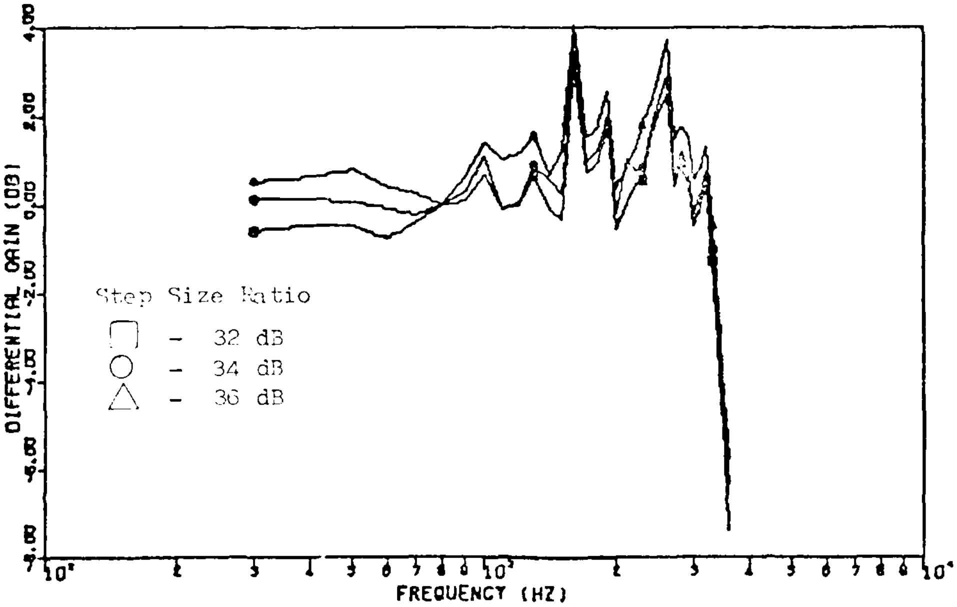

Figure 44 a. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Parameters Matched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

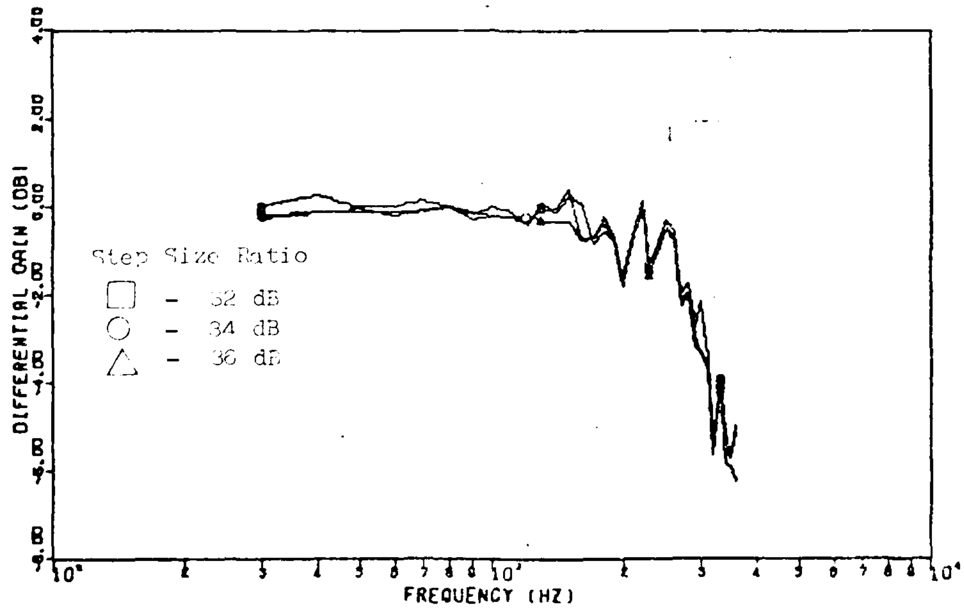

Figure 44 b. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Parameters Matched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

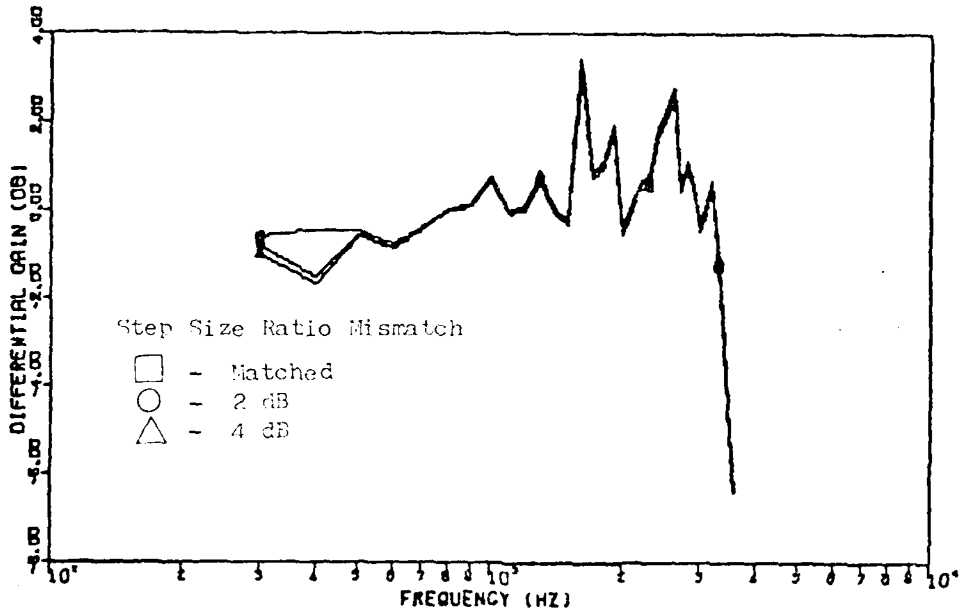

Figure 45 a. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Step Size Ratios Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

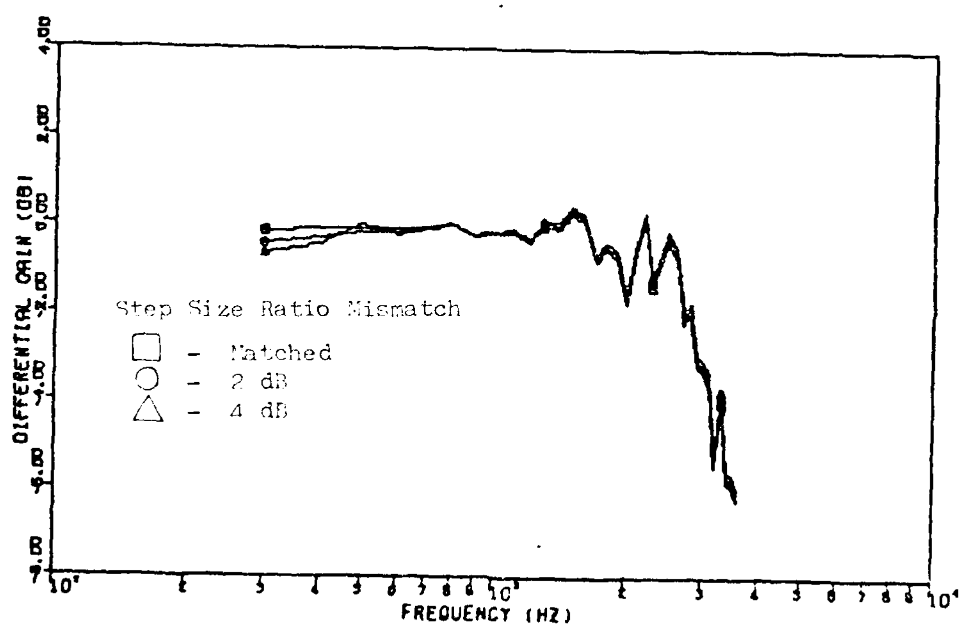

Figure 45 b. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Step Size Ratios Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

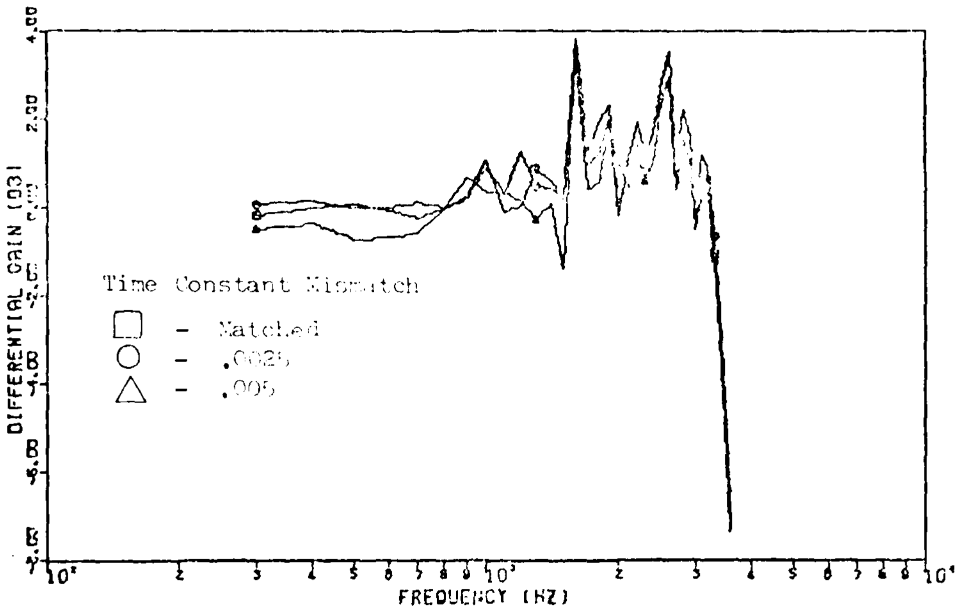

Figure 46a. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

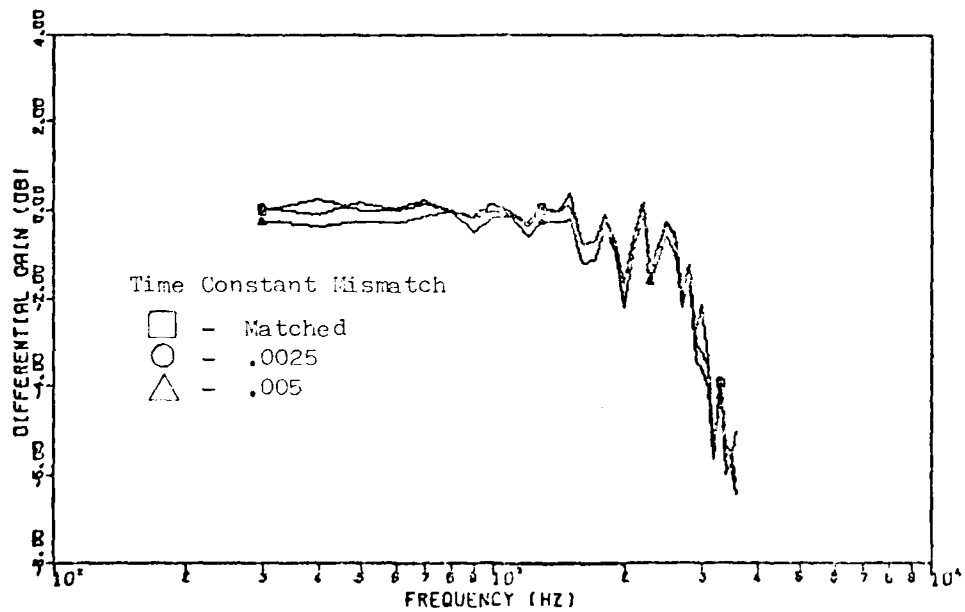

Figure 46b. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

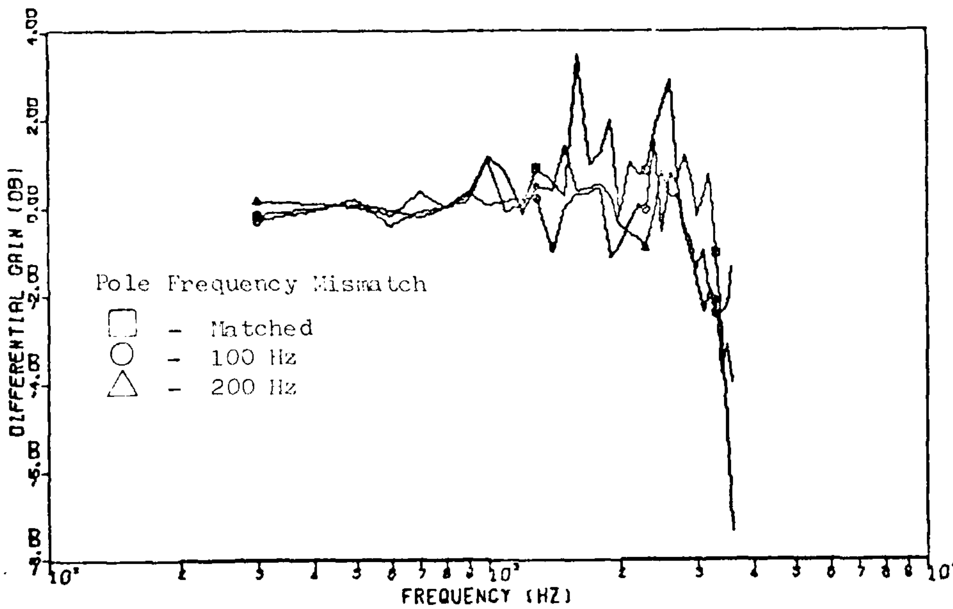

Figure 47a. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

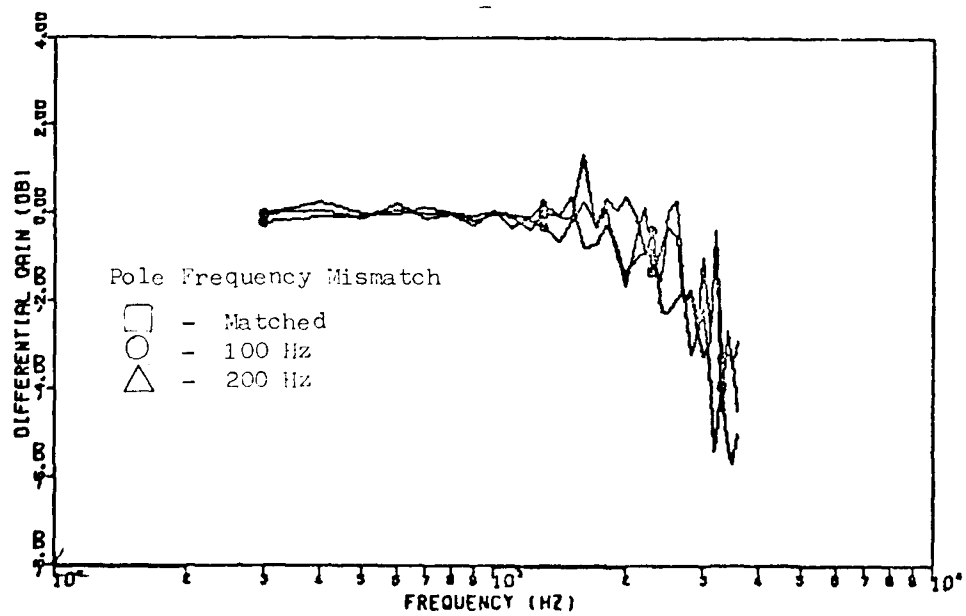

Figure 47b. CVSD System Gain Variation vs. Frequency with Encoder and Decoder Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

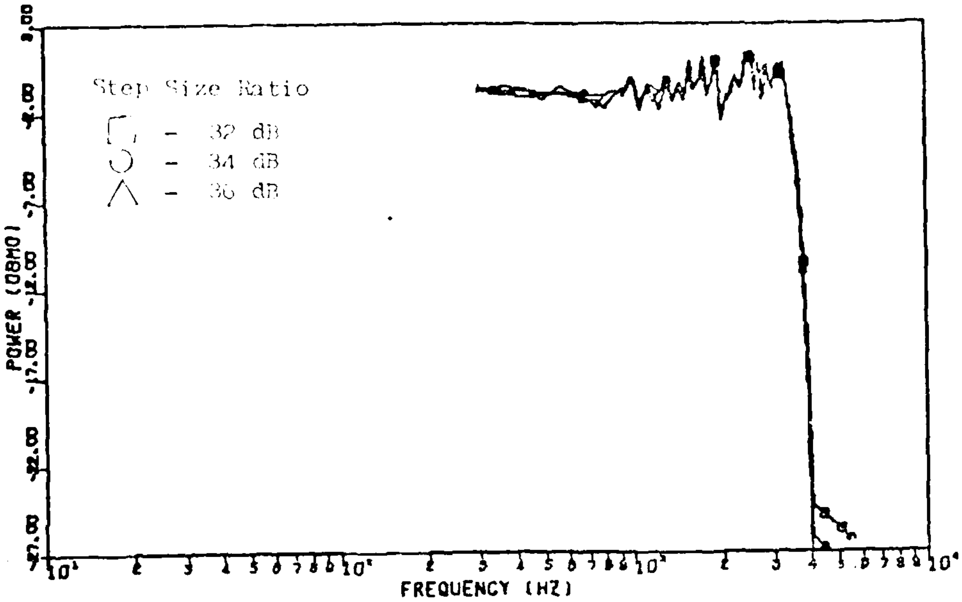

Figure 48 a. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Matched Parameters at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

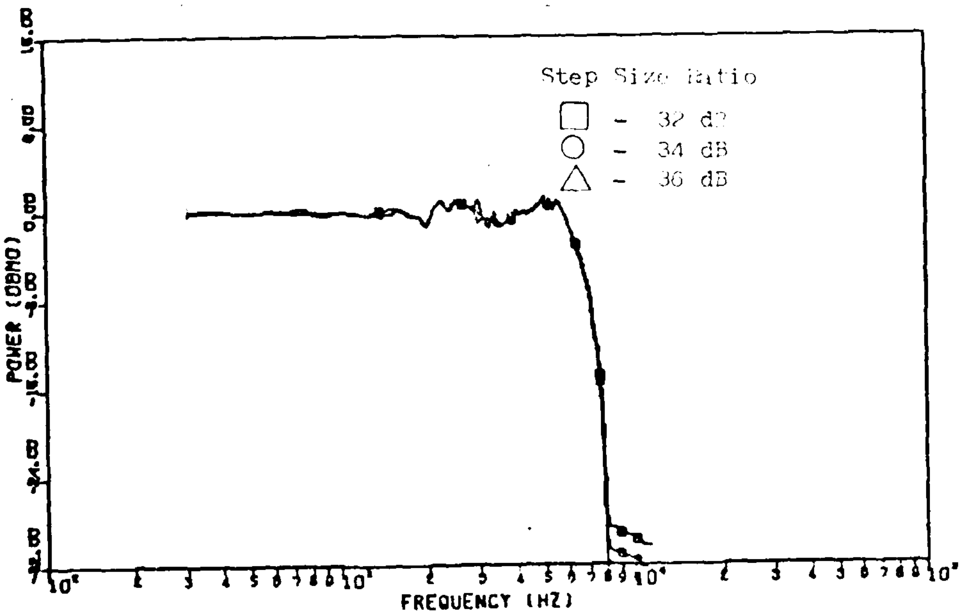

Figure 48 b. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Matched Parameters at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

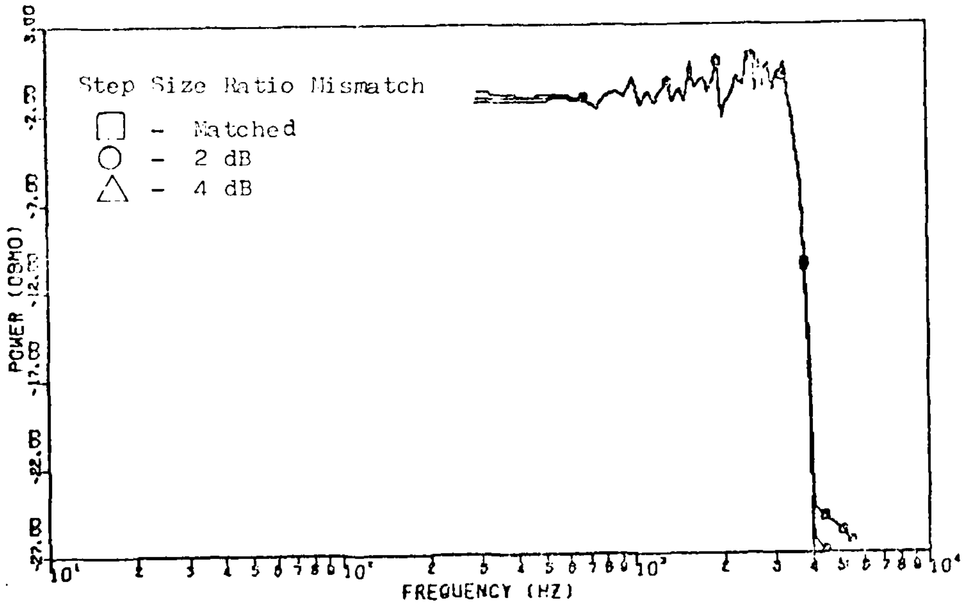

Figure 49a. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Step Size Ratios Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

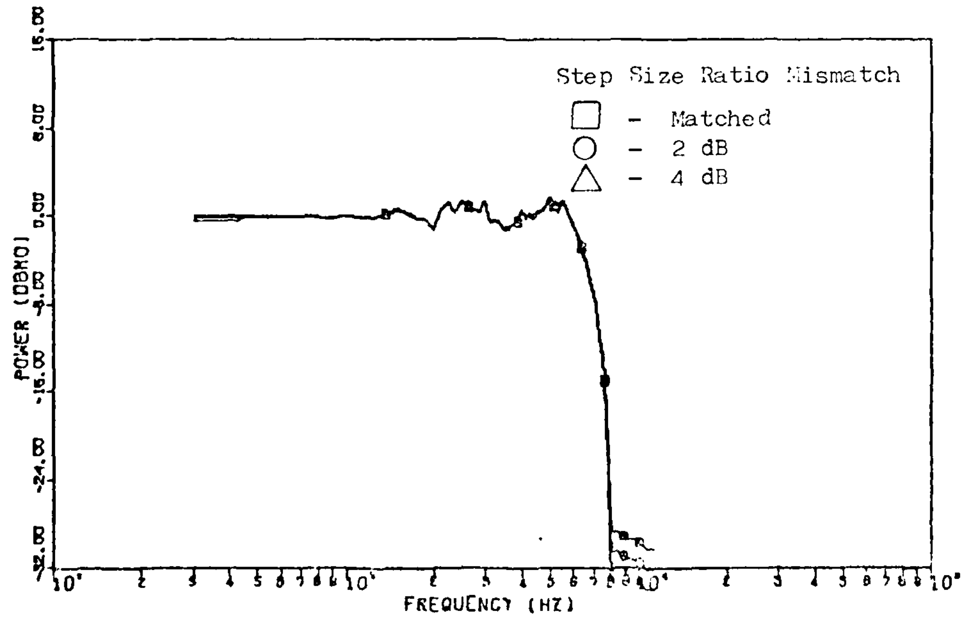

Figure 49b. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Step Size Ratios Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

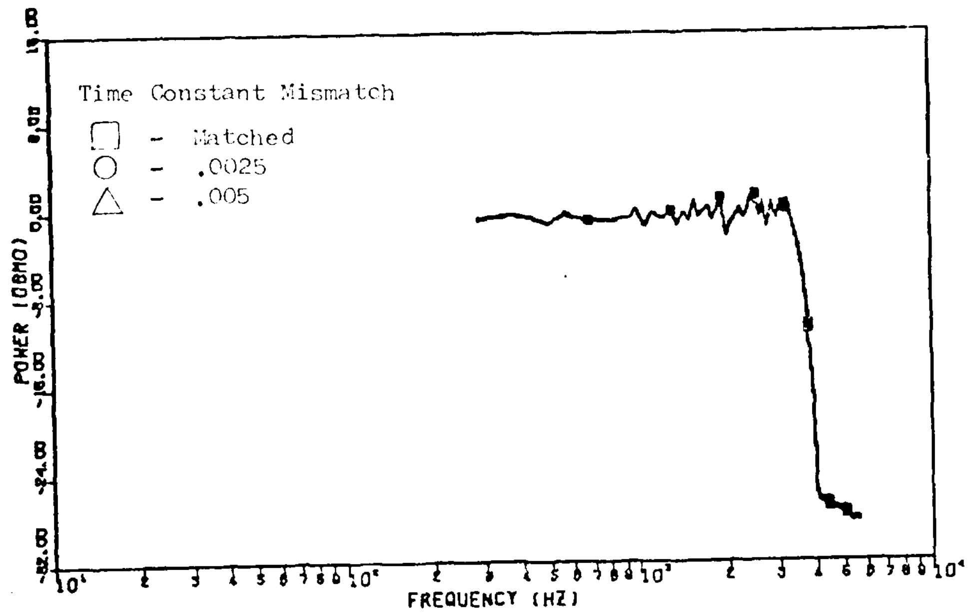

Figure 50 a. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Syllabic Filter Time Constants Mismatched at 16 kb/s Sample Rate (-20 dBmO Test Signal)

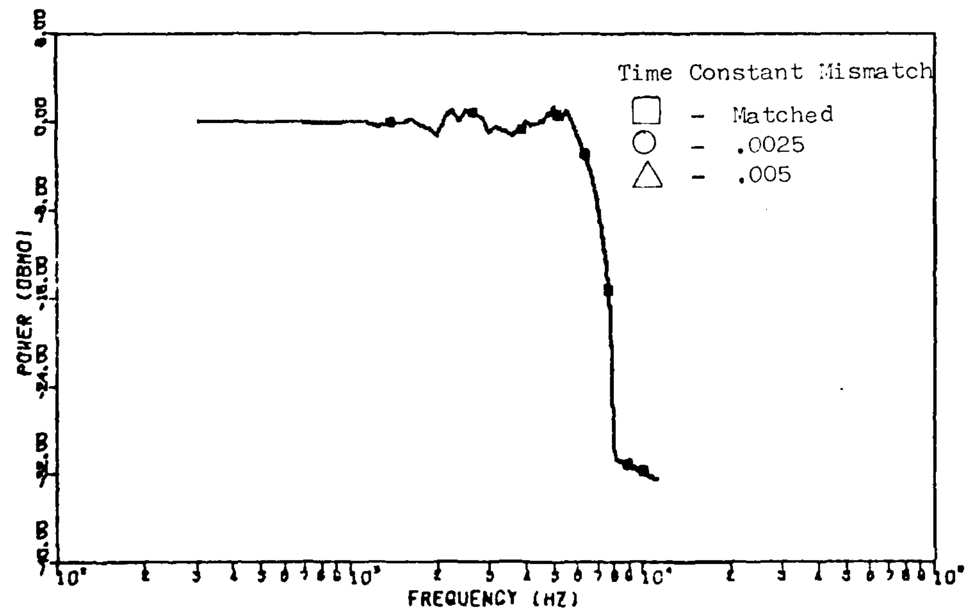

Figure 50 b. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Syllabic Filter Time Constants Mismatched at 32 kb/s Sample Rate (-20 dBmO Test Signal)

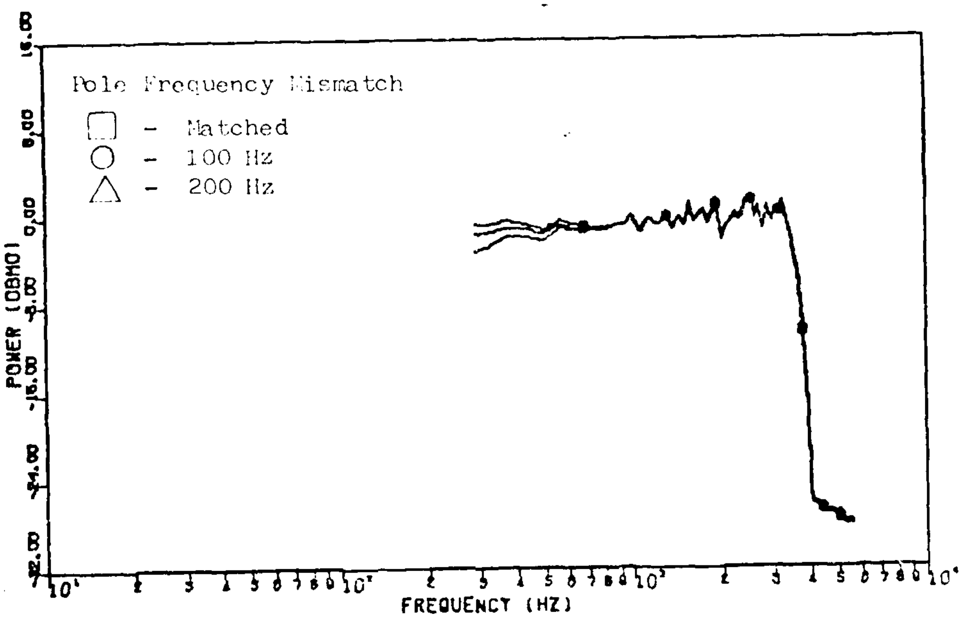

Figure 51a. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Primary Integrator Pole Frequencies Mismatched at 16 kb/s Sample Rate (-20 dBm0 Test Signal)

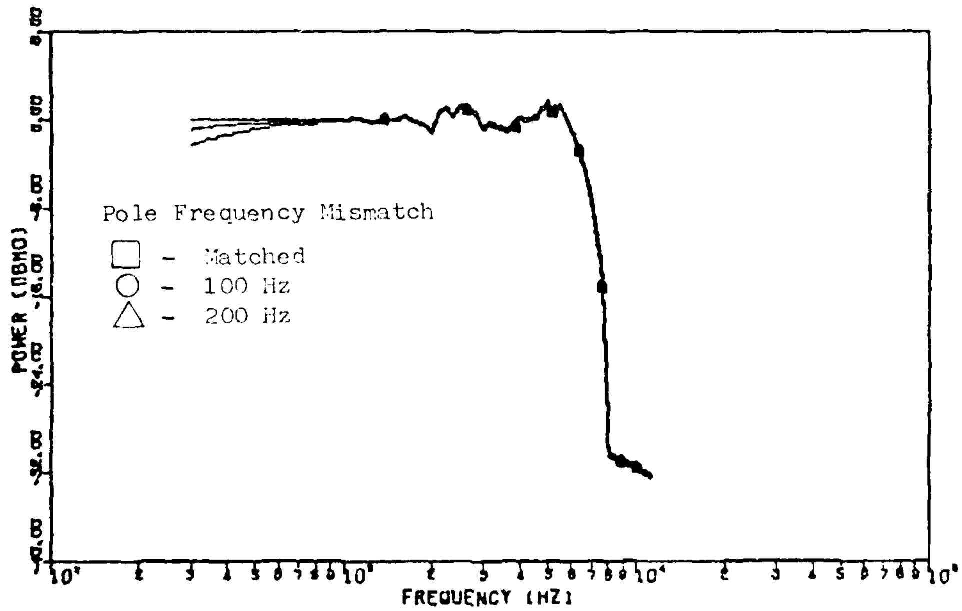

Figure 51b. CVSD Encoder/Decoder Back-to-Back Frequency Response Performance with Primary Integrator Pole Frequencies Mismatched at 32 kb/s Sample Rate (-20 dBm0 Test Signal)

V. Conclusions and Recommendations

The test results show that the computer model meets most of the performance criteria set by the draft standard when the system parameters are matched and at their nominal values. The output filter does not have the stop band loss characteristics specified in the standard and as a result, the system performance is marginal. Signal-to-noise ratios fall below the established criteria when the input power is less than -10 dBm0. In spite of this, the general effects of variations in the system parameter values and encoder/decoder mismatches can be observed in the test results.

1. When the encoder and decoder are matched, changes in the step size ratio, syllabic filter time constant, and the primary integrator pole frequency values within the tolerances allowed by the draft standard have a negligable effect on the transmitted signal.

2. If the step size ratio or the syllabic filter time constant are not the same in both the encoder and the decoder, the effect of the mismatch on the transmitted signal is negligable except at input power levels less than -32 dBm0. At these levels, the effects would not be noticeable to the system users.

3. System performance is most sensitive to encoder and decoder primary integrator pole frequency mismatches. All the performance tests show a larger deviation from the matched system performance when the primary integrators are mismatched. This type of parameter mismatch dominates mismatches of the other parameters.

4. The frequency response of the system is determined largely by the output filter when the pass band is restricted to less than ½ the sample rate. Above ½ the sample rate, the response is determined by the CVSD encoder and decoder pass band. The frequency selective measurement of the encoder/decoder response shows that the system is incapable of meeting the draft standard gain variation vs. frequency criteria given in figure 7a of Appendix A. The encoder and decoder alone have a response that falls off sharply above 4 kHz for the 16 kb/s sample rate, while the standard requires that the response not fall off more than 5 dB until 6 kHz is reached.

5. The specifications and tolerances given in the draft standard

appear adequate to assure reasonable system performance for voice signals. The transmitted signals will suffer some degradation due to parameter mismatches between the encoder and decoder. In most cases, the degradation is minimal, however, additional testing would be necessary to determine if the system response would continue to meet the draft standard criteria under mismatched conditions.

Recommendations

1. Since this model's performance is marginal under ideal conditions, parameter mismatch causes the performance to fall below the criteria set in the draft standard. Testing should be repeated using a filter that has higher stop band loss. The additional testing should concentrate on primary integrator response mismatches between the encoder and decoder, since the other parameters have little effect on the sytem response.

2. System tolerance to bit errors in the transmission system was not tested. A mismatch between the encoder and decoder may cause increased sensitivity to transmission errors. Testing to establish the system response to transmission bit error rate my be desireable.

3. The draft standard specifications for gain variation in the region between 4 kHz and 6 kHz for the 16 kb/s sample rate should be modified. Performance should be allowed to roll off as sharply as possible above 3.6 kHz.

4. Testing wss performed using continuous sinusoidal test signals only. If quasi-analog signals are expected to be used on the CVSD system, testing should be repeated using this type of signal at various input keying speeds. Since the CVSD algorithm depends on the high sample to sample correlation characteristics of the voice signal, quasi-analog signals can be expected to suffer more degradation as the result of encoder/decoder mismatches. More restrictive tolerances may be necessary to assure adequate performance with these signals. In addition, more standards may be necessary, such as delay distortion specifications.

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APPENDIX A

DRAFT STANAG

ON

THE ANALOGUE/DIGITAL CONVERSION OF
SPEECH SIGNALS FOR
TACTICAL, DIGITAL, AREA
COMMUNICATIONS SYSTEMS

JUNE 1978

INTRODUCTION

This STANAG is one of a series, which, when taken as a whole, will specify the necessary technical parameters to allow digital, tactical area communications systems to interface.

This particular STANAG specifies the analogue/digital conversion of signals in the voice band to result in a digital bit rate of either 16 or 32 kbit/s per second. (32 kbits/sec in the interrim period).

In order for two communications systems to interface they must use the same conversion process so that the speech signals may be reconstituted at the destination.

This STANAG specifies a delta coder/decoder (change in speech level is coded using syllabic companding controlled by a 3 bit logic.

AD-A100 781 AIR FORCE INST OF TECH WRIGHT-PATTERSON AFB OH SCHOO--ETC F/G 9/2
INVESTIGATION OF CONTINUOUSLY VARIABLE SLOPE DELTA MODULATOR/DE--ETC(U)
DEC 80 J A LERSCH

UNCLASSIFIED AFIT/GE/EE/80D-28 NL

2 OF 2
AD A100781

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1. General

1.1 Analogue/digital conversion of telephone signals (speech or other voice-band signals shall be performed by a delta coder/decoder using syllabic companding controlled by a three bit logic.

1.2 Block diagrams of the coder and decoder are shown in Figures 1 and 2.

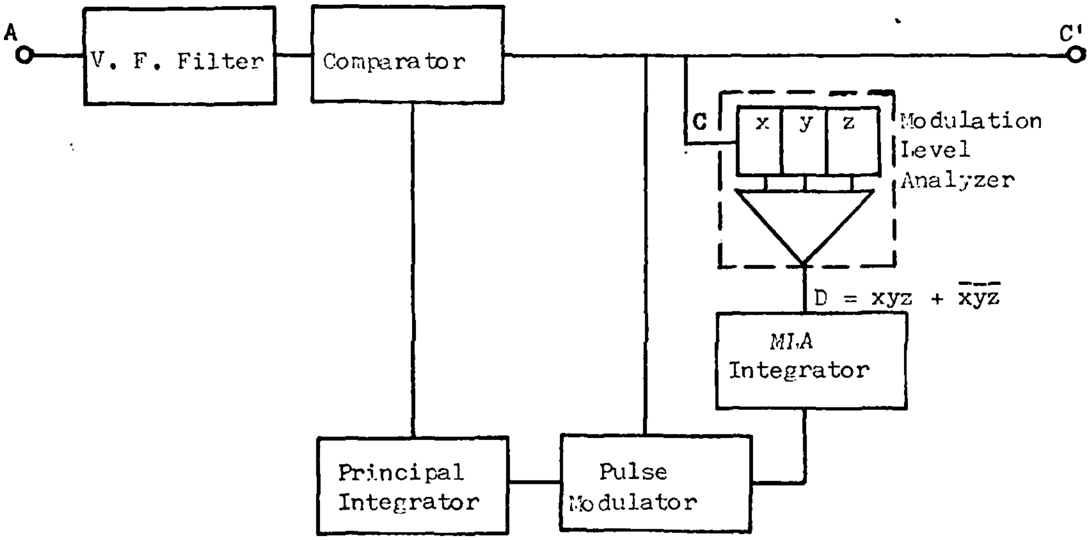

Fig. 1 - Block Schematic of the Coder

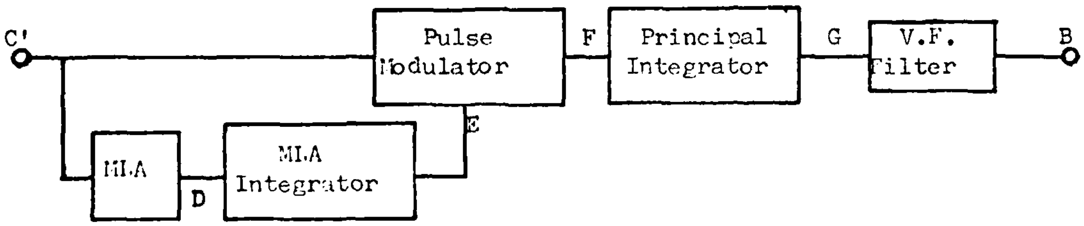

Fig. 2 - Block Schematic of the Decoder

2. Four-wire to Four-wire Audio Frequency Characteristics

2.1 Relative Level at Points A and B

The relative levles at points A and B shall be -4 dBr.

2.2 The absolute level is calculated by the equation dBm = dBr + dBm0.

2.3 Impedance at Points A and B

The nominal value of the impedance at points A and B shall be 600 ohms.

2.4 Return Ioss at Points A and B against 600 ohms

The return loss at points A and B shall be ≥ 16 dB in the frequency range from 300 Hz to 3400 Hz against a load resistor of 600 ohms with an input level of -20 dBm0.

2.5 Symmetry at Points A and B

Points A and B shall be balanced and not referred to ground, i.e. shall be floating.

3. Details of the Coder and Decoder Circuits

3.1 Input and Output Audio Filters

For frequencies above 6 kHz, each filter shall have an attenuation of ≥ 25 dB.

3.2 Frequency Response of the Principal Integrator

The ideal amplitude frequency characteristic between points F and G is shown in Figure 3.

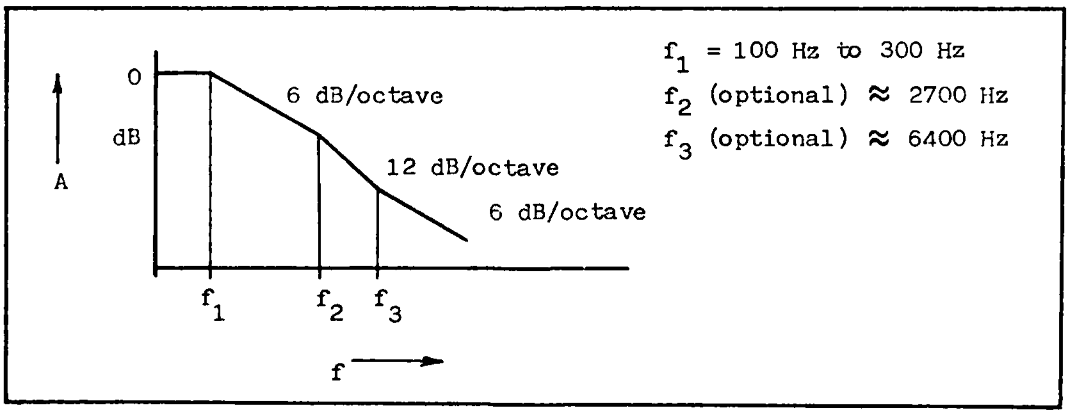

Fig. 3 - Ideal Amplitude Frequency Characteristic of the Principal Integrator

3.3 Modulation Level

A signal of 800 Hz and 0 dBm0, applied to point A of the coder shall give a duty cycle (mean proportion of binary '1' digits at point D each one indicating a run of 3 equal bits at point C) of c_d = 0.5 at point D of the modulation level analyzer MLA).

3.4 Compression and Expansion

In the coder and decoder the quantizing step size q which drives the principle integrator at Point F, shall have an essentially linear relationship to the duty cycle at point D of the MLA integrator (see Figure 4).

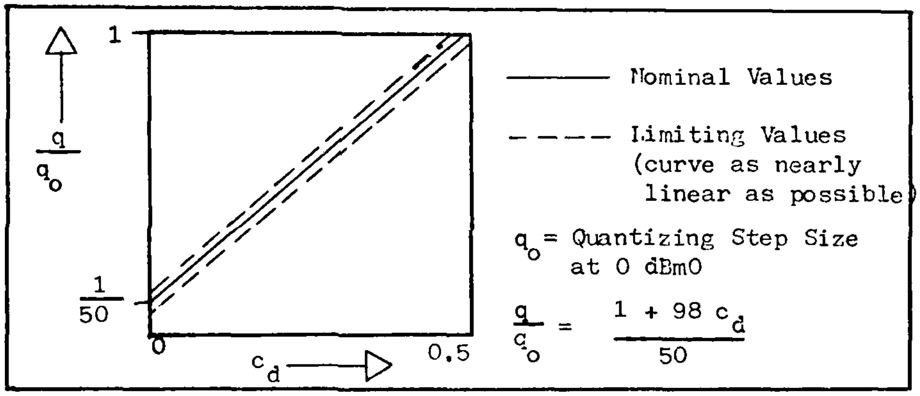

Fig. 4 - Relation between MLA Output Duty Cycle and Size of Quantizing Steps

It follows that the ratio of the quantizing step size at point F corresponding to a duty cycle of c_d = 0.5 at point D of the MLA integrator at the minimum step size q_o shall be 34 dB (provisional tolerance: ± 2dB).

3.5 Companding Speed

The following is valid for the condition that C is connected to C'. When an 800 Hz sine ave signal at point A is suddenly changed from -42 dBm0 to 0 dBm0 the output signal at point B shall reach 90% of its final value within 2 mS to 4 mS.

NOTE:-

The MLA integrator circuits of the coder and decoder shall have the same characteristics and hence the same companding speed.

3.6 Procedure for Testing the Delta Decoder

The test bit sequence generator is connected to the decoder input point C' (see Figure 2).

Testing is performed by means of periodical test bit sequences (listed in Table 1) which result in audio signals at 800 Hz at the decoder output point B. The 800 Hz levels at point B shall conform to the values given in Table 1.

When the signal at point C' is switched from the periodical test bit sequence to the periodical test bit sequence g, then the output signal at point B shall reach 90% of its final value within 5.5 mS to 11.5 mS. When the signal at point C' is switched from the periodical test bit sequence g to the periodical test bit sequence a, then the output signal at point B shall reach 10% of the value of the periodical test bit sequence g within 4 mS to 8mS.

NOTE:- for clarification

For an RC circuit in the MIA integrator with time constants of 4 mS for both charging and discharging, the envelope characteristic of the output signal at point B is shown in Figure 5. For the case of switching the signal at point C' from the sequence g to sequence a, the amplitude at the beginning of discharging is at the first moment after switching higher - by a factor of 50 - than the final value which is reached asymptotically. The final value equals -42 dBm0, i.e. 0.00794, the amplitude at the beginning of the discharging is hence 0.397 (c_d = 0). The value of 10% is then reached at 5.76 mS.

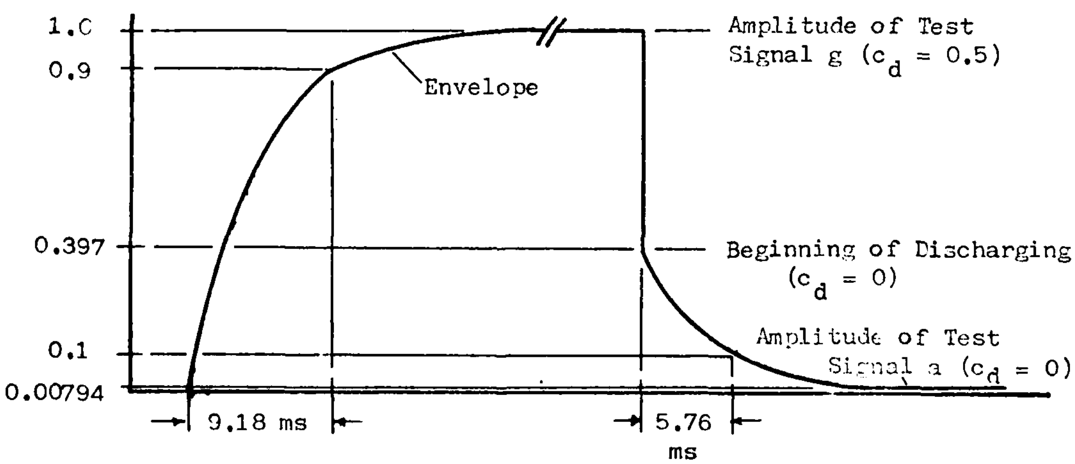

Fig. 5 - Envelope Characteristic of the Output Signal at Point B (Half the Envelope)

Table 1 - Bit Sequences for Testing Delta Decoder

Test Signals		Bit Sequence	c_d	(dBm0) x)
a	(1)	1011010010010010110	0	-41.5 ± 3
	(2)	1011011010101001001001001001010101101101		-42 ± 3
b	(1)	11011001001001001101	0.05	-25 ± 2
	(2)	1011011010101001001000100100101011011011		
c	(1)	10110101000100101011	0.1	-19 ± 2
	(2)	1101101101010010001000100100101011011101		-18.5 ± 2
d	(1)	11011001000010011011	0.2	-11 ± 2
	(2)	1101110110010100010000100010011010111011		-11.5 ± 2
e	(1)	11011010000010010111	0.3	-6.5 ± 1.5
	(2)	11101110110010001000000010001001101110111		
f	(1)	11011010000001001111	0.4	-3 ± 1.5
	(2)	1111011101010001000000001000101011101111		
g	(1)	11101010000000101111	0.5	0 ± 1
	(2)	1111101110100010000000000100010111011111		

c_d Duty cycle at point D of the modulation level analyzer (MLA)

(1) Sequence of 20 bits for a digit rate o 16 kbits/s

(2) Sequence of 40 bits for a digit rate o 32 kbits/s

x) For the relative level see para. 2.1 above.

4. Electrical Performance at Points A and B

4.1 General

The required values under 4.2 to 4.8 are valid for the condition that C is connected to C'.

For measurement, the input (point A) and the output (point B) are to be terminated with 600 ohms, and signals whose frequencies are sub-multiples of the sampling rate shall be avoided. Accordingly, where a nominal test signal frequency of 800 Hz is indicated, the actual frequency shall be slightly different; a preferred value is 820 Hz, but frequencies from 804 to 860 Hz.

The measurements according to Sections 4.2 to 4.5 shall be performed selectively.

4.2 Insertion Loss between Points A and B

The insertion loss between points A and B at 800 Hz with an input level of 0 dBm0 shall be 0 dBm0 $\pm$ 2 dB. The insertion loss contributed by the transmit and receive sides shall not exceed one-half of the value.

4.3 Attenuation Distortion with Frequency

The attenuation distortion relative to 800 Hz measured with an input level of -20 dBm0 applied to point A shall be within the limits of Figure 6. The distortion contributed by the transmit side alone, measured at point G of the coder, shall not exceed the limits indicated by the broken lines in Figure 6.

4.4 Variation of Gain with Input Level

The deviation of the output level compared with the value at -20 dBm0 shall not exceed the limits given in Figure 7 for a frequency of 800 Hz.

4.5 Idle Channel Noise

Idle channel noise at 16 kbits/s:

The idle channel noise at point B shall not exceed -45 dBm0p. The level of any single frequency, measured selectively, shall not exceed -50 dBm0 in the frequency range from 0.3 kHz to 8 kHz.

Idle channel noise at 32 kbit/s:-

The idle channel noise at point B shall not exceed -60 dBm0p. The level of any single frequency, meaured selectively, shall not exceed -65 dBm0 in the frequency range from 0.3 kHz to 16 kHz.

4.6 Variation of Quantization and Harmonic Distortion with Input Level

The distortion shall be mesured unweighted with a sinewave test signal at 800 Hz. With such a signal applied to point A, the ratio of signal to distortion power at the output point B shall be above the limits of Figure 8.

4.7 Variation of Quantizing and Harmonic Distortion with Frequency

The distortion shall be measured unweighted with a sinewave test signal of -20 dBm0. With such a test signal applied to point A, the ratio of signal to distortion power at the output point B shall be above the limits of Figure 9.

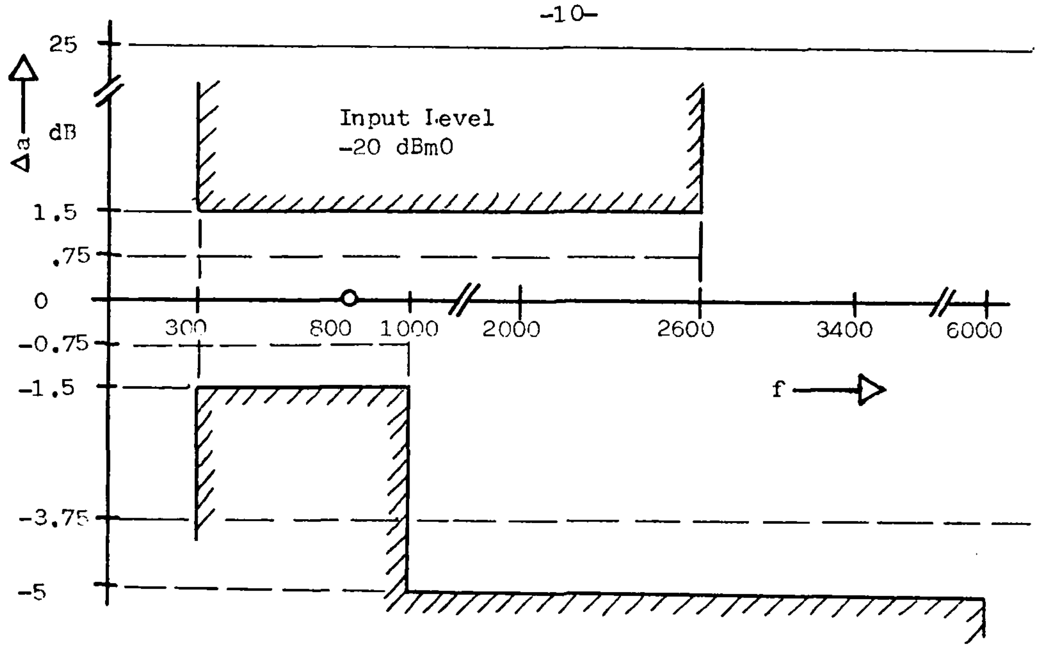

Fig. 6a - Attenuation Distortion with Frequency
at a Digit Rate of 16 kbit/s

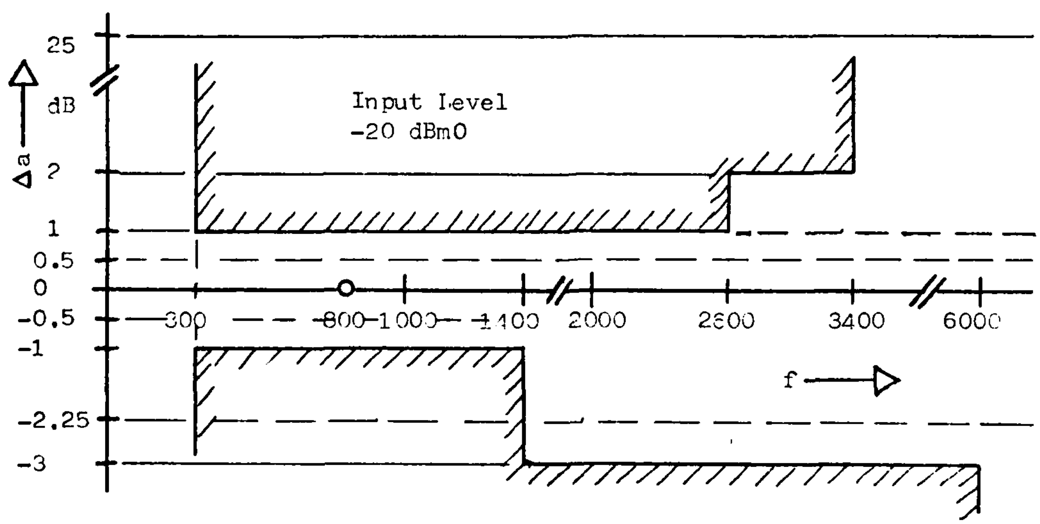

Fig 6b - Attenuation Distortion with Frequency
at a Digit Rate of 32 kbit/s

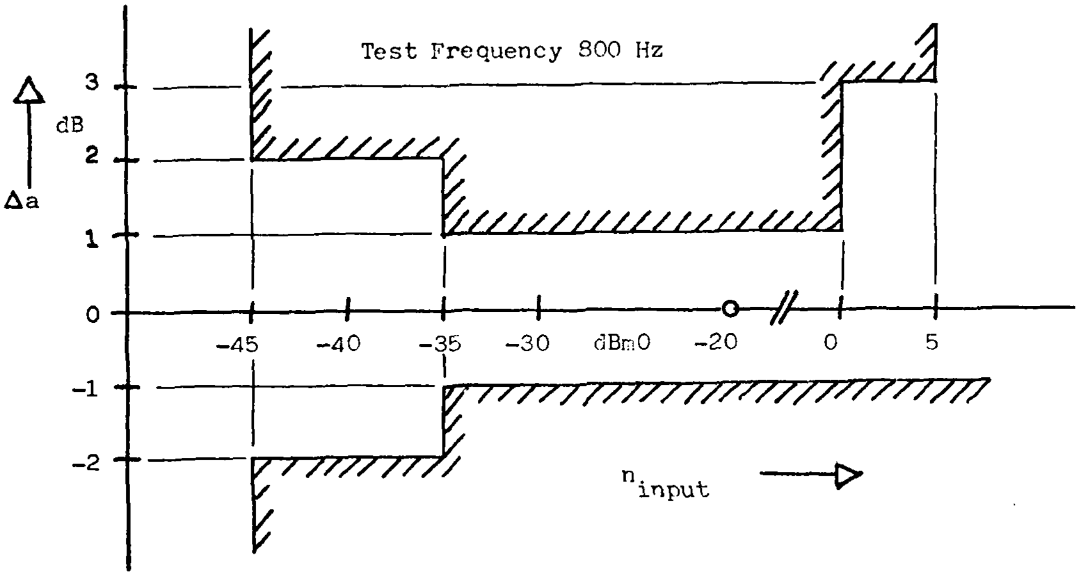

Fig. 7a - Variation of Gain with Input Level at a Digit Rate of 16 kbit/s

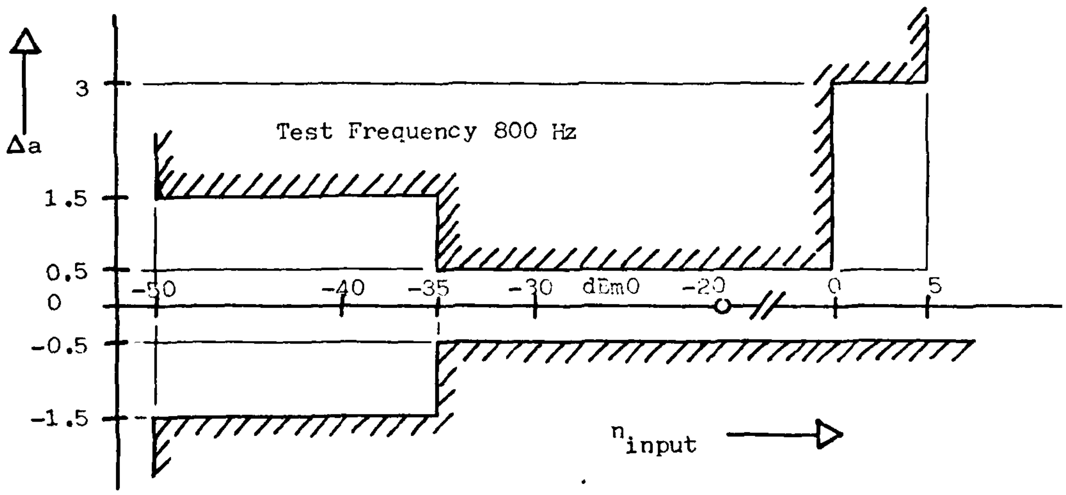

Fig. 7b - Variation of Gain with Input Level at a Digit Rate of 32 kbit/s

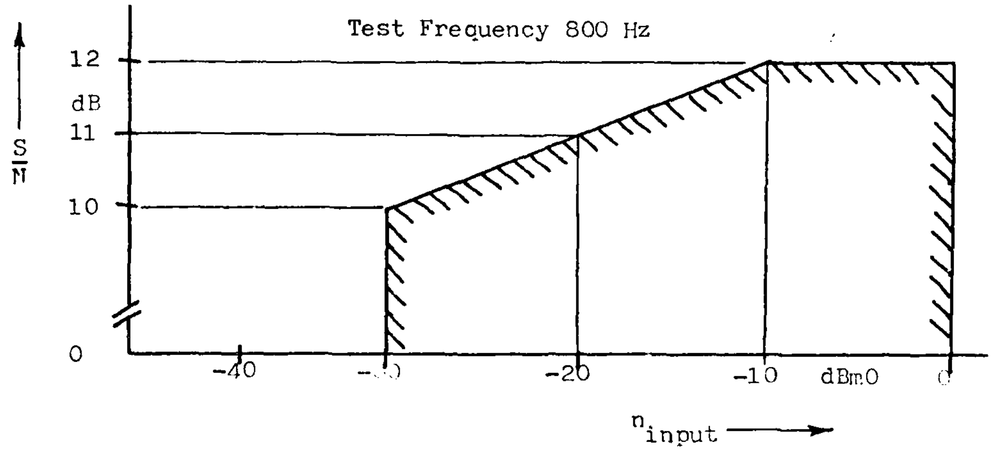

Fig. 8a - Quantizing and Harmonic Distortion with Level at a Digit Rate of 16 kbit/s

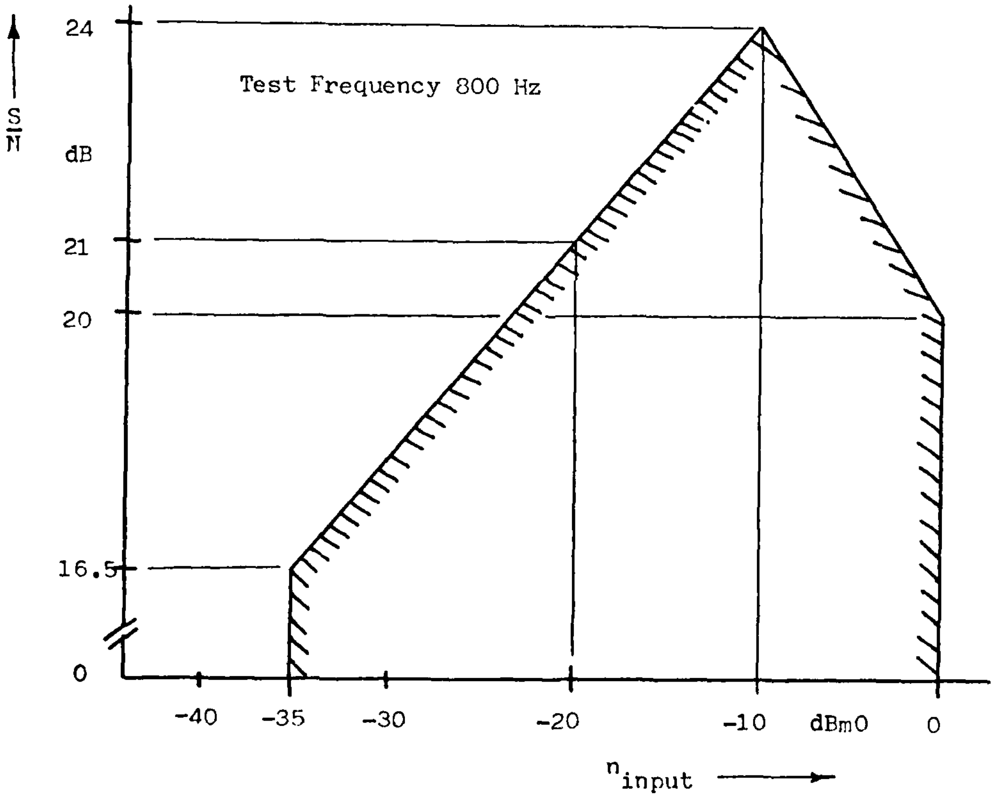

Fig. 8b - Quantizing and Harmonic Distortion with Level at a Digit Rate of 32 kbit/s

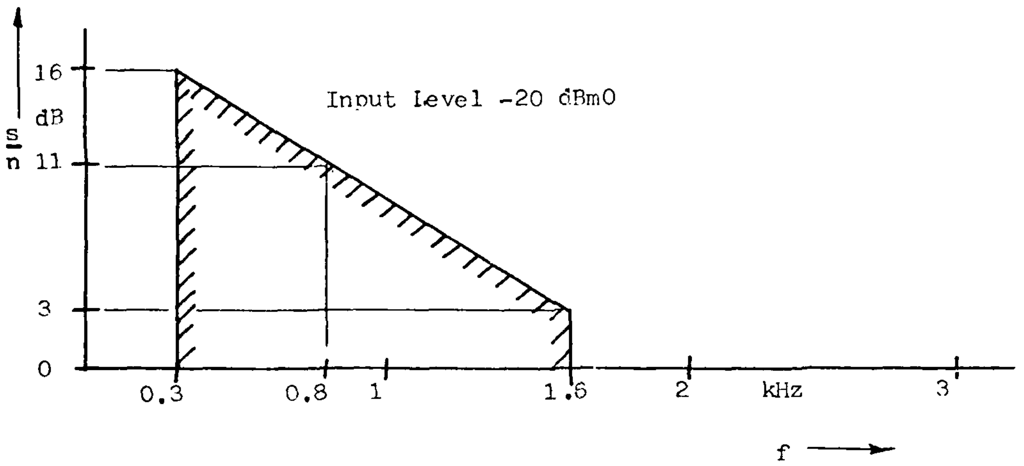

Fig. 9a - Quantizing and Harmonic Distortion with Frequency at a Digit Rate of 16 kbit/s

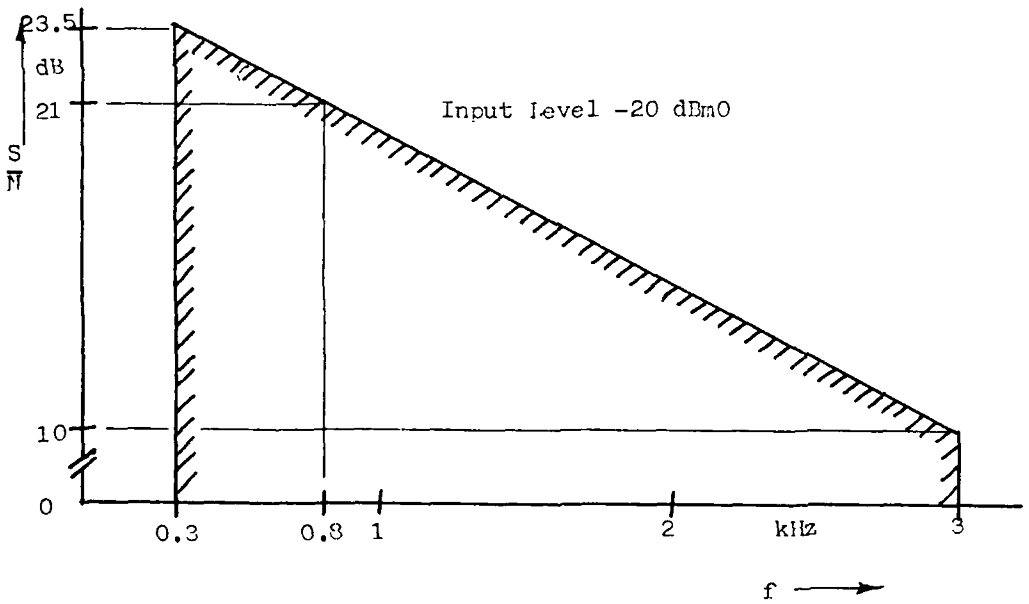

Fig. 9b - Quantizing and Harmonic Distortion with Frequency at a Digit Rate of 32 kbit/s

APPENDIX B

CVSD Encoding Subroutine

```
      SUBROUTINE ENCODE1(INPUT,OUTPUT,N,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)
C-----------------CVSD ENCODER SUBROUTINE--------------------------------
C     THE SUBROUTINE CONVERTS AN INPUT TIME FUNCTION TO AN OUTPUT BINARY
C     DATA STREAM.  BOTH INPUT AND OUTPUT IS DONE THROUGH ARRAYS.
C*********************** VARIABLES **************************************
C     INPUT = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES.
C     OUTPUT = AN ARRAY CONTAINING THE OUTPUT BINARY DATA STREAM.
C     N = THE NUMBER OF SAMPLES.
C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRATOR.
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTER.
C     ALPHA = THE DECAY RATE OF THE PRIMARY INTEGRATOR.
C     BETA = THE DECAY RATE OF THE SYLLABIC FILTER.
C     EN = THE SIGN OF THE DIFFERENCE BETWEEN THE CURRENT INPUT AND
C        THE CURRENT ESTIMATE.
C     EN1 = THE SIGN OF THE DIFFERENCE ONE TME PERIOD AGO.
C     EN2 = THE SIGN OF THE DIFFERENCE TWO TIME PERIODS AGO.
C     UMIN = THE MINIMUM INPUT TO THE SYLLABIC FILTER.
C     UMAX = THE MAXIMUM INPUT TO THE SYLLABIC FILTER.
C     FS = THE SAMPLE RATE.
C     DELTAN = THE CURRENT STEP SIZE.
C     DIF = THE DIFFERENCE BETWEEN THE CURRENT INPUT AND THE CURRENT
C        ESTIMATE.
C     XN = THE CURRENT ESTIMATE.
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR FOR THE
C        CURRENT INPUT STRING.
C************************************************************************
C************************************************************************
C-------------------SUBROUTINE START-------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      REAL INPUT(N)
      INTEGER OUTPUT(N)
      DATA XN/0/, EN1/0/, EN2/0/, PI/3.1415926536/
      SUM = 0.
      DELTAN = UMIN
C---- CALCULATE DECAY RATES OF ENCODER FILTERS

      ALPHA = EXP (-(2. * PI * FC1 / FS))
      BETA = EXP (-(2. * PI / TC / FS))
C---- START ENCODING
      DO 50 I = 1,N
C---- CALCULATE THE OUTPUT OF THE COMPARATOR
      DIF= INPUT(I) - XN
      EN = SIGN(1.,DIF)
C---- GENERATE NEW ESTIMATE
      XN = ALPHA * XN + (1 - ALPHA) * DELTAN *EN
      U = UMIN
```

```
C---- GENERATE THE NEXT OUTPUT OF THE SLOPE OVERLOAD DETECTOR

      IF ((((EN .AND. EN1) .AND. EN2) .EQ. 1.) .OR.
     1(((EN .AND. EN1) .AND. EN2) .EQ. -1.)) V = VMAX
      IF (V .EQ. VMAX) SUM = SUM + 1

C---- GENERATE NEXT STEP SIZE

      DELTAN = BETA * DELTAN + (1 - BETA) * V

C---- SHIFT THE SLOPE OVERLOAD DETECTOR SHIFT REGISTER

      EN2 = EN1
      EN1 = EN
      OUTPUT(I) = EN

C---- POLAR TO BINARY CONVERT

      IF (EN .EQ. -1) OUTPUT(I) = 0
50    CONTINUE

C---- CALCULATE SLOPE OVERLOAD DETECTOR DUTY CYCLE

      DC = SUM / N

      RETURN
      END
```

APPENDIX C

CVSD Decoding Subroutine

```
      SUBROUTINE DECODE1(INPUT,OUTPUT,N,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)
C---------------------CVSD DECODING SUBROUTINE----------------------
C     THIS SUBROUTINE DECODES THE BINARY DATA STREAM CONTAINED IN THE
C     INPUT ARRAY AND PUTS THE OUTPUT TIME FUNCTION SAMPLES IN THE OUT-
C     PUT ARRAY.
C************************** VARIABLES ******************************
C     INPUT = AN ARRAY CONTAINING THE INPUT BINARY DATA STREAM.
C     OUTPUT = AN ARRAY CONTAINING THE OUTPUT TIME FUNCTION
C     N = THE NUMBER OF SAMPLES
C     FS = THE SAMPLE RATE
C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRATOR
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTER.
C     XN = THE CURRENT OUTPUT TIME SAMPLE
C     EN1 = THE SIGN OF THE DIFFERENCE ONE TIME PERIOD AGO
C     EN2 = THE SIGN OF THE DIFFERENCE TWO TIME PERIODS AGO.
C     UMAX = THE MAXIMUM INPUT TO THE SYLLABIC FILTER
C     UMIN = THE MINIMUM INPUT TO THE SYLLABIC FILTER
C     DELTAN = THE CURRENT STEP SIZE
C     ALPHA = THE DECAY RATE OF THE PRIMARY INTEGRATOR
C     BETA = THE DECAY RATE OF THE SYLLABIC FILTER
C     DC = THE SLOPE OVERLOAD DETECTOR DUTY CYCLE.
C******************************************************************
C******************************************************************
C---------------------SUBROUTINE START------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION INPUT(N), OUTPUT(N)
      DATA XN/0/, EN1/0/, EN2/0/, PI/3.1415926536/
      SUM = 0.
      DELTAN = UMIN
C---- CALCULATE FILTER DECAY RATES
      ALPHA = EXP (-(2. * PI * FC1 / FS))
      BETA = EXP (-(2. * PI / TC / FS))
C---- START DECODING
      DO 50 I = 1,N

C---- GET NEXT INPUT BIT AND BINARY TO POLAR CONVERT
      EN = INPUT(I)
      IF (INPUT(I) .EQ. 0) EN = -1
C---- GENERATE NEXT OUTPUT TIME SAMPLE
      XN = ALPHA * XN + (1 - ALPHA) * DELTAN *EN
      U = UMIN
```

```
C---- GENERATE THE NEXT OUTPUT OF THE SLOPE OVERLOAD DETECTOR

      IF ((((EN .AND. EN1) .AND. EN2) .EQ. 1.) .OR.
     1(((EN .AND. EN1) .AND. EN2) .EQ. -1.)) V = VMAX
      IF (V .EQ. VMAX) SUM = SUM + 1

C---- GENERATE NEXT STEP SIZE

      DELTAN = BETA * DELTAN + (1 - BETA) * V

C---- SHIFT THE SLOPE OVERLOAD DETECTOR SHIFT REGISTER

      EN2 = EN1
      EN1 = EN
      OUTPUT(I) = XN
50    CONTINUE

C---- CALCULATE THE SLOPE OVERLOAD DETECTOR DUTY CYCLE

      DC = SUM / N
      RETURN
      END
```

APPENDIX D

FIR Filtering Subroutine (FILTER)

```
      SUBROUTINE FILTER(XT, N, NP, B)
C-----------------------FIR FILTER SUBROUTINE-------------------------
C     THIS SUBROUTINE FILTERS AN INPUT TIME FUNCTION SAMPLE STRING USING
C     FILTER COEFFICIENTS GENERATED BY AN EXTERNAL FILTER GENERATOR
C     ROUTINE.  THE FILTERRED TIME FUNCTION SAMPLES ARE PLACED IN THE
C     SAME ARRAY AS THE INPUT AND RETURN TO THE CALLING PROGRAM.  DUE
C     TO THE FILTER TECHNIQUE, 200 SAMPLES AT THE BEGINNING AND END
C     OF THE SAMPLE STRING ARE LOST.
C************************** VARIABLES ********************************
C     XT = THE ARRAY CONTAINING THE INPUT SAMPLE STRING AND AFTER PRO-
C        CESSING, THE FILTERRED SAMPLE STRING.
C     N = THE NUMBER OF SAMPLES IN THE INPUT ARRAY
C     NP = THE NUMBER OF FILTER COEFFICIENTS
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS
C*********************************************************************
C-----------------------SUBROUTINE START------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION XT(N), B(NP)
C---- FILTER THE INPUT SAMPLE STRING
      DO 100 I = 1,4600
      K = 200 + I
      DO 50 J = 1,NP
      IF (J .EQ. 1) SUM = B(J) * XT(K)
      SUM = SUM + B(J) * (XT(K + J - 1) + XT(K - J + 1))
50    CONTINUE
      XT(I) = SUM
100   CONTINUE
      RETURN
      END
```

APPENDIX E

FIR Filter Coefficient Generating Subroutine

```
      SUBROUTINE FLTRGEN(BETA,GAMMA,NP,B)
C-------------------MAXIMALLY FLAT FILTER PROGRAM------------------
C
C     THIS PROGRAM OUTPUTS THE FIR FILTER COEFFICIENTS CALCULATED BY
C      SUBROUTINE MXFLAT.  THIS ROUTINE EITHER RETURNS THE CALCULATED
C     COEFFICIENTS TO THE CALLING PROGRAM OR PRINTS OUT THE ERROR MES-
C     SAGES WHEN THE COEFFICIENTS CANNOT BE DETERMINED DUE TO THE CHOICE
C     OF INPUT PARAMETERS.
C
C     THIS SUBROUTINE AND THE MXFLAT AND RATPRX SUBROUTINES USED TO
C     GENERATE THE MAXIMALLY FLAT FIR FILTER COEFFICIENTS ARE ADAPTED
C     FROM A PROGRAM DEVELOPED BY J. F. KAISER OF BELL LABORATORIES.
C     THIS PROGRAM WAS PUBLISHED IN 'PROGRAMS FOR DIGITAL SIGNAL PRO-
C     CESSING' BY THE IEEE PRESS.
C
CXXXXXXXXXXXXXXXXXXXXXXX VARIABLES XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
C
C     BETA = THE NORMALIZED CENTER FREQUENCY OF THE TRANSITION BAND
C
C     GAMMA = THE NORMALIZED WIDTH OF THE TRANSITION BAND
C
C     NP = THE NUMBER OF FILTER COEFFICIENTS
C
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS
C
C     LIMIT = THE LARGEST NUMBER OF FILTER COEFFICIENTS ALLOWED
C
C     IERR = THE NUMBER OF THE ERROR MESSAGE
C
C     A & C = WORKING ARRAYS
C
CXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
C
C----------------------SUBROUTINE START---------------------------
C
C---- INITIALIZE VARIABLES AND ARRAYS
C
      DIMENSION A(200),B(200),C(200)
      LIMIT = 200
      CALL MXFLAT(BETA, GAMMA, NP, A, B, C, LIMIT, IERR)
C
C---- PRINT RESULTS
C
      IF (IERR .GT. 1) WRITE(6,9998) BETA, GAMMA
9998  FORMAT(' FOR BETA = ',F5.3,' AND GAMMA = ',F5.3)
      GO TO (10, 20, 30, 40), IERR
10    RETURN
20    WRITE(6,9997)
9997  FORMAT(' BETA NOT IN RANGE 0. - .5')
      STOP
30    WRITE(6,9996)
9996  FORMAT(' GAMMA NOT IN RANGE')
      STOP
40    WRITE(6,9995)
9995  FORMAT(' GAMMA TOO SMALL, MIN IS .04+')
      STOP
      END
```

APPENDIX F

Subroutine MXFLAT - Part of FIR Filter Generator

```
      SUBROUTINE MXFLAT(BE, GA, NP, A, B, C, LIMIT, IERR)
C------------------SUBROUTINE MXFLAT-----------------------------
C
C     THIS SUBROUTINE COMPUTES THE COEFFICIENTS OF A MAXIMALLY FLAT FIR
C     LINEAR PHASE FILTER.
C
C************************** VARIABLES ****************************
C
C     BE = CENTER OF THE TRANSITION REGION, RANGE = 0. TO .5
C        FREQUENCY IS NORMALIZED TO THE SAMPLE RATE.
C
C     GA = WIDTH OF THE TRANSITION REGION, WHERE THE OUTPUT AMPLITUDE
C        DECREASES FROM 95% TO 5%.
C
C     LIMIT = THE MAXIMUM NUMBER OF COEFFICIENTS IN THE FILTER
C
C     B = THE ARRAY CONTAINING THE FILTER COEFFICIENTS
C
C     IERR = ERROR MESSAGES
C           1, NORMAL RETURN
C           2, BETA NOT IN RANGE
C           3, GAMMA NOT IN RANGE
C           4, GAMMA TOO SMALL, LESS THAN .04
C
C     A = WORKING ARRAY
C
C     C = WORKING ARRAY
C
C     K = NUMBER OF ZEROS AT NYQUIST FREQUENCY
C
C     L = NUMBER OF ZERO DERIVATIVES AT ZERO FREQ
C
C     NT = FILTER HALF ORDER = NP - 1
C
C******************************************************************
C
C----------------------SUBROUTINE START---------------------------
C
C---- INITIALIZE VARIABLES AND ARRAYS
C
      DIMENSION A(LIMIT), B(LIMIT), C(LIMIT)
      IERR = 1
      NP = 0
      TWOPI = 8. * ATAN(1.0)
      IF ((BE .LE. 0.) .OR. (BE .GE. .5)) GO TO 80
      BM = AMIN1(2. * BE, 1. - 2. * BE)
      IF ((GA .LE. 0.) .OR. (GA .GE. BM)) GO TO 90
      NT = INT(1. / (4. * GA * GA))
      IF (NT .GT. 160) GO TO 100
      AC = (1. + COS(TWOPI * BE)) / 2.
      QLIM = LIMIT
      CALL RATPRX(AC, NT, K, NP, QLIM)
      N = 2 * NP - 1
      IF (K .EQ. 0) K = 1
C
C---- COMPUTE MAGNITUDE AT NP POINTS
C
      C(1) = 1.
```

```
      A(1) = 1.
      LL = NT - K
      L = LL + 1
      DO 40 I = 2,NP
      FF = FLOAT(I-1)/FLOAT(N)
      C(I) = COS(TWOPI * FF)
      X = (1. - C(I)) / 2.
      SUM = 1.
      IF (K .EQ. NT) GO TO 40
      Y = X
      DO 30 J = 1,LL
      FJ = J
      JL = K - 1
      Z = Y
      IF (K .EQ. 1) GO TO 20
      DO 10 JJ = 1,JL
      AJ = JJ
      Z = Z * (1. + FJ / AJ)
10    CONTINUE
20    Y = Y * X
      SUM = SUM + Z
30    CONTINUE
      A(I) = SUM * (1. - X) ** K
40    CONTINUE
C---- CALUCLATE WEIGHTING COEFS BY AN N-POINT IDFT
      DO 70 I = 1,NP
      B(I) = A(1) / 2.
      DO 60 J = 2,NP
      M = MOD((I-1) * (J-1),N)
      IF (M .LE. NT) GO TO 50
      M = N - M
50    B(I) = B(I) + C(M+1) * A(J)
60    CONTINUE
      B(I) = 2. * B(I) / FLOAT(N)
70    CONTINUE
      RETURN
80    IERR = 2
      RETURN
90    IERR = 3
      RETURN
100   IERR = 4
      RETURN
      END
```

APPENDIX G

Subroutine RATPRX - Part of FIR Filter Generator

```
      SUBROUTINE RATPRX(A, N, K, NP, QLIM)
C----------------------SUBROUTINE RATPRX-------------------------------
C
C     THIS SUBROUTINE COMPUTES THE RATIONAL FRACTION APPROXIMATION, K/NP
C        TO NUMBER A WITHIN THE LIMIT OF N <= NP <= 2*N FOR THE DENOMIN-
C        ATOR.
C************************* VARIABLES **********************************
C     A = THE DESIRED NUMBER
C     N = INTEGER MAX LOWER LIMIT ON NP
C     K = INTEGER NUMERATOR
C     NP = INTEGER DENOMINATOR
C     N RETURNS AS NP - 1
C     K / NP IS NEAREST TO A IN THE ALGEBRAIC SENSE, N < LIMIT
C**********************************************************************
C--------------------SUBROUTINE START----------------------------------
      IF (N .LE. 0) GO TO 3
      AA = ABS(A)
      AI = IFIX(AA)
      AF = AMOD(AA,1.)
      QMAX = 2 * N
      IF (QMAX .GT. QLIM) QMAX = QLIM
      Q = N-1
      EM = 1.
1     Q = Q + 1.
      IF (Q .GT. QMAX) GO TO 2
      PS = Q * AF
      IP = PS + .5
      E = ABS((PS - FLOAT(IP)) / Q)
      IF ( E .GE. EM) GO TO 1
      EM = E
      PP = IP
      QQ = Q
      GO TO 1
2     K = SIGN(AI * QQ + PP, A)
      NP = QQ
      N = NP - 1
      IF (K .EQ. NP) GO TO 4
      RETURN
3     K = 0
      N = -1
      NP = 0
      RETURN
4     NP = QMAX
      K = NP - 1
      N = K
      RETURN
      END
```

APPENDIX H

Syllabic Filter Output vs. Slope Overload Detector Duty Cycle

```
      PROGRAM STEPSZ(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT,PLOT)
C
C-------------------------STEP SIZE CALCULATION-------------------------
C
C      THIS PROGRAM CALCULATES THE STEP SIZE OUTPUT OF THE SYLLABIC FIL-
C      TER USED IN THE CVSD ENCODER AND DECODER.  THE STEP SIZE DETERMIN-
C      ED AS A FUNCTION OF THE SLOPE OVERLOAD DETECTOR OUTPUT DUTY CYCLE.
C      THE DETECTOR DUTY CYCLE IS VARIED FROM 0 TO .5 AND THE AVERAGE
C      SYLLABIC FILTER OUTPUT AMPLITUDE IS CALCULATED AND PLOTTED
C      THE CALCULATIONS ARE PERFORMED FOR BOTH 16 AND 32 KB/S SAMPLE
C      RATES AND THE DATA PLOTTED ON THE SAME GRAPH.
C
C
C************************* VARIABLES **********************************
C
C
C      CD = A REAL ARRAY CONTAINING THE DUTY CYCLE POINTS AT WHICH CALCU-
C         LATION OF THE AVERAGE STEP SIZE IS PERFORMED.
C
C      STEP = A REAL ARRAY CONTAINING THE AVERAGE STEP SIZES
C
C      VMAX = A REAL VARIABLE USED AS THE INPUT TO THE SYLLABIC FILTER
C         TO DETERMINE THE MAXIMUM STEP SIZE.
C
C      VMIN = A REAL VARIABLE USED AS THE INPUT TO THE SYLLABIC FILTER
C         TO DETERMINE THE MINIMUM STEP SIZE.
C
C      FS = THE SAMPLE RATE BEING USED.
C
C      TC = THE TIME CONSTANT OF THE SYLLABIC FILTER.
C
C      BETA = THE DECAY RATE OF THE OUTPUT OF THE SYLLABIC FILTER DURING
C         ONE TIME STEP.
C
C      I, J = COUNTING INDICES FOR THE 'DO' LOOPS.
C
C      DELTA = THE CURRENT VALUE OF THE STEP SIZE.
C
C      SUM = A RUNNING TOTAL OF THE STEPS CALCULATED TO BE USED FOR
C         AVERAGING.
C
C      V = THE CURRENT VALUE OF THE INPUT TO THE SYLLABIC FILTER.
C
C      JDI, JT = INTEGER VARIABLES USED IN CALCULATING WHEN THE INPUT TO
C         THE SYLLABIC FILTER SHOULD BE CHANGED FROM VMIN TO VMAX.  THE
C         INPUT IS VMAX FOR 'ITH' 'J'.
C
C**********************************************************************
C
C------------------PROGRAM START---------------------------------------
C
C---- INITIALIZE VARIABLES AND ARRAYS
C
      DIMENSION CD(52), STEP(52)
      DATA PI/3.1415926538/
      DO 1000 IT = 1,2
C
C---- ENTER WORKING VARIABLES
C
      READ *, FS, TC,FC1, RATIO
C
C---- CALCULATE SYLLABIC FILTER DECAY RATE
C
      BETA = EXP (-(2. * PI /TC / FS))
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
C
C---- START CALCULATION OF STEP SIZES
C
      DO 100 I = 2,50
      DELTA = 0.
      SUM = 0.
C
C---- CALCULATE 500 STEP VALUES FOR EACH STEP OF DUTY CYCLE
C
      DO 50 J = 1,500
C
C---- THE INPUT TO THE SYLLABIC FILTER IS VMIN UNLESS J IS EVENLY
C     DIVISABLE BY I.
C
      V = VMIN
      JDI = J / I
      JT = JDI * I
      IF (JT .EQ. J) V = VMAX
      DELTA = BETA * DELTA + (1. - BETA) * V
```

```
      SUM = SUM + DELTA
50    CONTINUE
C
C---- CALCULATE DUTY CYCLE AND AVERAGE STEP SIZE
C
      CD(I-1) = 1. / I
      STEP(I-1) = SUM / 500.
100   CONTINUE
C
C---- CALCULATE THE AVERAGE STEP VALUE FOR A DUTY CYCLE OF ZERO.
C
      DELTA = 0.
      SUM = 0.
      V = VMIN
      DO 200 J = 1,500
      DELTA = BETA * DELTA + (1. - BETA) * V
      SUM = SUM + DELTA
200   CONTINUE
      CD(50) = 0.
      STEP(50) = SUM / 500.
C
C---- PRINT AND PLOT THE RESULTS.

C
      PRINT *, ' FOR THE FOLLOWING SYSTEM PARAMETERS:'
      PRINT *,'      SAMPLE RATE = ',FS,' BPS'
      PRINT *,'      TC = ',TC
      PRINT *, '      FC1 = ',FC1
      PRINT *, '      RATIO = ',RATIO
      PRINT *,' THE FOLLOWING SYSTEM PARAMETERS  ARE:'
      PRINT *,' VMAX = ',VMAX,'      VMIN = ',VMIN
      PRINT *,' MAXIMUM STEP SIZE = ',STEP(1),'   MINIMUM STEP SIZE = ',
     1STEP(50)
      IF (IT .GT. 1) GO TO  900
      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)
      CALL SCALE(CD,10.0,50,1)
      CALL SCALE(STEP,6.0,50,1)
      CALL AXIS(0.0,0.0,13HSTEP SIZE (V),13,6.0,90.0,STEP(51),STEP(52))
      CALL AXIS(0.0,0.0,10HDUTY CYCLE,-10,10.0,0.0,CD(51),CD(52))
      CALL RECT(0., 0., 6., 10., 0., 3)
900   CONTINUE
      ICHAR = IT - 1
      CALL LINE(CD,STEP,50,1,10,ICHAR)
1000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX I

VMAX Calculating Subroutine

```
      SUBROUTINE VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
C*********************************************************************
C     THIS SUBROUTINE CALCULATES THE VALUE OF VMAX AND VMIN BASED ON
C     THE INPUT SAMPLE RATE, SYLLABIC FILTER TC, PRIMARY INTEGRATOR
C     ROLL-OFF FREQUENCY (FC1), AND THE RATIO BETWEEN THE MAXIMUM STEP
C     SIZE AND MINIMUM STEP SIZE OUTPUT OF THE SYLLABIC FILTER.  THIS
C     CALCULATION IS PERFORMED AT A REFERENCE FREQUENCY OF 800 HZ AND
C     SIGNAL AMPLITUDE OF 0 DBM0.  THE VALUES OF VMAX AND VMIN ARE
C     CALCULATED SUCH THAT THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR
C     OUTPUT IS .5.
C************************** VARIABLES ********************************
C     VMAX = THE MAXIMUM INPUT TO THE SYLLABIC FILTER
C     VMIN = THE MINIMUM INPUT TO THE SYLLABIC FILTER
C     FS = THE SAMPLE RATE
C     FC1 = THE ROLL-OFF FREQUENCY OF THE PRIMARY INTEGRATOR
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTER
C     RATIO = THE RATIO BETWEEN THE MAXIMUM STEP SIZE AND THE MINIMUM
C        STEP SIZE OUTPUT OF THE SYLLABIC FILTER
C     TS = AN ARRAY CONTAINING THE TEST SIGNAL SAMPLES
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE ENCODER
C     PEAK1 = THE PEAK VALUE OF THE TEST SIGNAL AMPLITUDE
C     DIF = THE DIFFERENCE BETWEEN THE SLOPE OVERLOAD DETECTOR DUTY
C        CYCLE AND THE DESIRED VALUE OF .5.
C     RAT = THE RATIO OF THE DUTY CYCLE DIFFERENCE TO THE DESIRED VALUE
C********************* SUBROUTINES USED ******************************
C     SIGNAL = THE TEST SIGNAL GENERATOR
C     VMINOPT = CALCULATES THE VALUE OF VMIN THAT PAIRS WITH THE CAL-
C        CULATED VALUE OF VMAX
C     ENCODE1 = THE CVSD ENCODER
C*********************************************************************
C---------------------SUBROUTINE START--------------------------------
C---- INITIALIZE ARRAYS AND VARIABLES
      DIMENSION TS(4096)
      INTEGER BINOUT(4096)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      PEAK1 = A(0.)
      VMAX = 10.
C---- CALCULATE TEST SIGNAL SAMPLES
      CALL SIGNAL(TS,4096,FS,800.,0.,PEAK1,0.)
C---- START VMAX CALCULATION LOOP
5     CONTINUE
C---- CALCULATE ESTIMATED ENCODER PARAMETERS
      CALL VMINOPT(VMAX,VMIN,FS,TC,RATIO)
C---- PROCESS THE TEST SIGNAL
      CALL ENCODE1(TS,BINOUT,4096,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
C---- FIND THE DIFFERENCE BETWEEN THE DUTY CYCLE USING THE ESTIMATED
C     VMAX AND VMIN AND THE DESIRED DUTY CYCLE
      DIF = DC - .5
      RAT = DIF / .5
```

```
C---- IF THE DUTY CYCLE IS WITHIN 1% OF THE DESIRED VALUE RETURN
C     THE VMAX AND VMIN VALUES TO THE CALLING PROGRAM

      IF (ABS(RAT) .LE. .01) GO TO 900

C---- OTHERWISE REESTIMATE VMAX AND REPEAT CALCULATIONS

      VMAX = VMAX + .5 * RAT * VMAX
      GO TO 5
900   CONTINUE
      RETURN
      END
```

APPENDIX J

VMIN Calculating Subroutine

```
      SUBROUTINE VMINOPT(VMAX,VMIN,FS,TC,RATIO)
C*************************************************************************
C
C     THIS SUBROUTINE CALCULATES THE VALUE OF VMIN THAT PAIRS WITH
C     THE VALUE OF VMAX THAT IS INPUT SO THAT THE RATIO OF THE MAX-
C     IMUM STEP SIZE TO MINIMUM STEP SIZE AT THE OUTPUT OF THE SYLLABIC
C     FILTER IS WITHIN .01% OF THE VALUE SPECIFIED.
C
C************************** VARIABLES ***********************************
C
C     VMAX = THE MAXIMUM INPUT VALUE OF THE SYLLABIC FILTER
C
C     VMIN = THE MINIMUM VALUE INPUT TO THE SYLLABIC FILTER
C
C     FS = THE SAMPLE RATE
C
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTER
C
C     MAXSTEP = THE MAXIMUM STEP SIZE AT THE OUTPUT OF THE SYLLABIC FIL-
C        TER
C
C     MINSTEP = THE MINIMUM STEP SIZE AT THE OUTPUT OF THE SYLLABIC FIL-
C        TER
C
C     BETA = THE DECAY RATE OF THE SYLLABIC FILTER
C
C     SUM = THE RUNNING SUM OF STEP SIZES
C
C     RATIO = THE DESIRED RATIO BETWEEN THE MAXIMUM STEP SIZE AND THE
C        MINIMUM STEP SIZE AT THE OUTPUT OF THE SYLLABIC FILTER IN DB
C
C     R = THE VOLTAGE RATIO EQUIVALENT OF RATIO
C
C     DELTA = THE CURRENT STEP SIZE
C
C*************************************************************************
C
C---- INITIALIZE VARIABLES AND ARRAYS
C
      REAL MAXSTEP, MINSTEP
      DATA PI/3.14159[illegible]536/
      R = 10. ** (RATIO / 20.)
C
C---- CALCULATE SYLLABIC FILTER DECAY RATE
C
      BETA = EXP (-(2. * PI / TC / FS))
C
C---- ESTIMATE INITIAL VALUE OF VMIN
C
      VMIN = VMAX / 100.
C
C     START CALCULATION LOOP
C
5     CONTINUE
C
C---- INITIAL RUNNING SUM AND STEP SIZE
C
      SUM = 0.
      DELTA = 0.


C---- CALCULATE AVERAGE MAXIMUM STEP SIZE
C
      DO 10 I = 1,500
      IDT = I / 2
      IT = IDT * 2
      V = VMIN
      IF (IT .EQ. I) V = VMAX
      DELTA = BETA * DELTA + (1. - BETA) * V
      SUM = SUM + DELTA
10    CONTINUE
      MAXSTEP = SUM / 500.
C
C---- REINITIALIZE RUNNING SUM AND STEP SIZE
      DELTA = 0.
      SUM = 0.
```

```
C---- CALCULATE CURRENT ESTIMATE OF THE MINIMUM STEP SIZE

      DO 15 I = 1,500
      DELTA = BETA * DELTA + (1. - BETA) * UMIN
      SUM = SUM + DELTA
15    CONTINUE
      MINSTEP = SUM / 500.

C---- FIND THE DIFFERENCE BETWEEN THE ESTIMATE AND THE SPECIFIED RATIO

      TMIN = MAXSTEP / R
      DIF = TMIN - MINSTEP
      RAT = ABS (DIF) / TMIN * 100.

C---- IF THE DIFFERENCE IS LESS THAN .01% RETURN UMIN

      IF (RAT .LE. .01) GO TO 999

C---- IF THE DIFFERENCE IS GREATER, THEN REESTIMATE UMIN AND REPEAT
C     CALCULATIONS

      UMIN = UMIN + .5 * DIF
      GO TO 5
999   CONTINUE
      RETURN
      END
```

APPENDIX K

CVSD System Step Response Program

```
      PROGRAM PULSE(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)

C     THIS PROGRAM PLOTS THE OUTPUT SIGNAL OF THE CVSD TRANSMISSION
C     SYSTEM WHEN THE INPUT SIGNAL IS A 800 HERTZ SINE WAVE
C     THAT VARIES IS AMPLITUDE FROM -42 DBM0 TO 0 DBM0.  THE TEST
C     SIGNAL GENERATOR ALTERNATELY GENERATES 500 SAMPLES AT -42 DBM0
C     AND 0 DBM0 IN ORDER TO DEMONSTRATE THE SYSTEM'S STEP RESPONSE
C     CHARACTERISTICS.  THE SYSTEM UNDER TEST CONSISTS OF THE INPUT
C     FILTER,THE CVSD ENCODER AND DECODER, AND THE OUTPUT FILTER.

C************************* VARIABLES ***********************************

C     TSIN = AN ARRAY CONTAINING THE INPUT TIME SERIES SAMPLES.

C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME SERIES
C        SAMPLES, THEN THE OUTPUT TIME SERIES SAMPLES OF THE FIR FILTER.

C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.

C     TIME = AN ARRAY CONTAINING THE TIME THAT THE FIRST 200 SAMPLES
C        ARE TAKEN SO THAT THEY MAY BE PLOTTED.

C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER

C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.

C     FS = THE SAMPLE RATE.

C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.

C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.

C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.

C     BETA = THE NORMALIZED CENTER OF THE TRANSITION BAND OF THE LOW
C        PASS FILTER.

C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.

C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.

C     NP = THE NUMBER OF FILTER COEFFICIENTS.

C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.

C     RATIO = THE RATIO BETWEEN THE MAXIMUM STEP SIZE AND THE MINIMUM
C        STEP SIZE IN DB

C**********************************************************************
C******************** SUBROUTINES USED *********************************

C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.

C     FACTOR, PLOT, AXIS, SCALE, RECT, LINE, PLOTE = CALCOMP PLOTTING
C        ROUTINES

C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.

C     ENCODE1 = THE CVSD ENCODER SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.

C     DECODE1 = THE CVSD DECODING SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.

C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME SERIES SAMPLES
C        USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.

C     PLTIME = A LINEAR PLOTTING ROUTINE TO PLOT SIGNAL AMPLITUDE VS.
C        TIME.

C     VMAXOPT = GENERATES VMAX AND VMIN FOR THE CVSD ENCODER AND DECODER
C        SUBROUTINES

C**********************************************************************
C**********************************************************************
```

```
C------------------PROGRAM START-----------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION TSIN(5000),TSOUT(5000),TIME(2000),B(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      AMP1 = -42.
      AMP2 = 0.
      PEAK1 = A (AMP1)
      PEAK2 = A (AMP2)
      KN = 0.

C---- INPUT AND PRINT THE WORKING VARIABLES

      READ *,  FS
      READ *,FC1, TC, RATIO
      READ *,BETA, GAMMA
      PRINT *, ' TRANSIENT RESPONSE TEST AT ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC,' AND RATIO = ',RATIO
      PRINT *,' FILTER PARAMETERS ARE, BETA = ',BETA,', GAMMA = ',GAMMA

C---- GENERATE FILTER COEFFICIENTS AND CVSD SYSTEM PARAMETERS

      CALL FLTRGEN(BETA,GAMMA,NP,B)
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)

C---- INITIALZE PLOTTER

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)

C---- GENERATE INPUT TIME FUNCTION SAMPLES

      CALL SIGNAL2(TSIN,5000,FS,600.,0.,PEAK1,PEAK2)

C---- FILTER THE INPUT

      CALL FILTER(TSIN,5000,NP,B)

C---- PROCESS THE INPUT TIME SERIES THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,4600,NP,B)

C---- PLOT THE  OUTPUT SIGNAL

      DO 5 I = 1,2000
      TIME(I) = I / FS
5     CONTINUE
      DO 6 I = 1,225
      IK = 2050 + I
      TSOUT(I) = TSOUT(IK)
6     CONTINUE
      CALL PLTIME(TIME,TSOUT,225,227)
      CALL PLOTE(N)
      END
C***********************************************************************
C***********************************************************************

      SUBROUTINE PLTIME(X, Y, N, NX)

C--------------------TIME VS. AMPLITUDE PLOTTER-------------------------

C     THIS SUBROUTINE MAKES A LINEAR PLOT OF TIME VS. AMPLITUDE.

C*************************** VARIABLES *********************************

C     X = THE ARRAY CONTAINING THE ORDINATE VALUES

C     Y = THE ARRAY CONTAINING THE ABSCISSA VALUES

C     N = THE NUMBER OF VALUES IN THE X AND Y ARRAYS

C     NX = N + 2

C*********************** SUBROUTINES USED ******************************

C     SCALE, AXIS,RECT, PLOT, PLOTE, LINE = CALCOMP PLOTTING ROUTINES

C***********************************************************************
```

```
C--------------------SUBROUTINE START--------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION X(NX), Y(NX)

C---- SCALE THE X AND Y ARRAYS

      CALL SCALE(X, 10., N, 1)
      CALL SCALE(Y, 6., N, 1)

C---- BOX IN THE PLOT

      CALL RECT(0., 0., 6., 10., 0., 3)

C---- DRAW THE AXES

      CALL AXIS(0., 0., 10HTIME (SEC), -10, 10., 0., X(N+1), X(N+2))
      CALL AXIS(0., 0., 13HAMPLITUDE (V), 13, 6., 90., Y(N+1), Y(N+2))

C---- PLOT THE POINTS

      CALL LINE(X, Y, N, 1, 0, 0)
      RETURN
      END

C*********************************************************************
C*********************************************************************

      SUBROUTINE SIGNAL2(OUTPUT,N,FS,FREQ1,FREQ2,AMP1,AMP2)

C*********************************************************************

C     THIS SUBROUTINE GENERATES A SINGLE FREQUENCY SINUSOIDAL SIGNAL
C     THAT ALTERNATELY HAS 500 SAMPLES AT ONE AMPLITUDE THEN 500 SAMPLES
C     AT A SECOND AMPLITUDE.

C********************** VARIABLES ************************************

C     OUTPUT = THE ARRAY CONTAINING THE OUTPUT TIME FUNCTION SAMPLES

C     N = THE NUMBER OF SAMPLES TO BE PRODUCED

C     FS = THE SAMPLE RATE

C     FREQ1 = THE FREQUENCY OF THE TEST SIGNAL

C     FREQ2 = UNUSED

C     AMP1 = THE AMPLITUDE OF THE FIRST 500 SAMPLES

C     AMP2 = THE AMPLITUDE OF THE SECOND 500 SAMPLES

C     AMP = THE SIGNAL AMPLITUDE CURRENTLY BEING USED

C*********************************************************************

C------------------------SUBROUTINE START----------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION OUTPUT(N)
      DATA PI/3.1415926536/
      K = 0
5     CONTINUE

C---- SET CURRENT SIGNAL AMPLITUDE

      AMP = AMP1
      AMP1 = AMP2
      AMP2 = AMP
      J = 0
10    CONTINUE

C---- GENERATE 500 TIME FUNCTION SAMPLES

      J = J + 1
      IF (J .GT. 500) GO TO 5
      K = K + 1

C---- IF N SAMPLES HAVE BEEN GENERATED, STOP PROGRAM

      IF (K .GT. N) GO TO 999
      OUTPUT(K) = AMP * SIN (2. * PI * FREQ1 / FS * K)
      GO TO 10
999   CONTINUE
      RETURN
      END
```

APPENDIX L

Idle Channel Noise Program

```
      PROGRAM IDLENOI(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C-------------------IDLE CHANNEL NOISE PROGRAM-----------------------
C     THIS PROGRAM MEASURES THE IDLE CHANNEL NOISE OF THE CVSD TRANS-
C     MISSION SYSTEM WHEN THE ENCODER AND DECODER ARE CONNECTED BACK-
C     TO-BACK AND THE INPUT TO THE ENCODER IS GROUNDED.
C     THE SYSTEM GAIN IS ADJUSTED SO THAT AN 800 HZ INPUT SIGNAL AT
C     -20 DBM0 PRODUCES A -20 DBM0 SIGNAL AT THE OUTPUT OF THE DE-
C     CODER.
C
C************************* VARIABLES **********************************
C     FREQ1 = THE REFERENCE FREQUENCY USED TO SET THE SYSTEM GAIN.
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES.
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME FUNCTION
C        SAMPLES, THEN THE OUTPUT TIME FUNCTION SAMPLES OF THE FIR FIL-
C        TER
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C     AMP1 = THE AMPLITUDE OF THE REFERENCE SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE.
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C     BETA = THE NORMALIZED 3 DB FREQUENCY OF THE OUTPUT FILTER.  THE
C        FREQUENCY IS NORMALIZED TO THE SAMPLE RATE.
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C     PIN = THE POWER OF THE INPUT SIGNAL IN DBM0.
C     POUT = THE POWER OF THE OUTPUT SIGNAL IN DBM0.
C     ICN = THE CALCULATED IDLE CHANNEL NOISE IN DBM0.
C     GAIN = THE VOLTAGE AMPLIFICATION OF THE SYSTEM.
C
C*********************************************************************
C********************* SUBROUTINES USED ******************************
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C     SIGNAL = THE TEST SIGNAL GENERATOR
C     ENCODE1 = THE CVSD ENCODER
C     DECODE1 = THE CVSD DECODER
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME FUNCTION SAM-
C        PLES USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME FUNC-
C        TION WITH IMPEDENCE = 600 OHMS.
C
C*********************************************************************
C*********************************************************************
```

```
C------------------PROGRAM START------------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION TSIN(5000), TSOUT(5000),  B(200)
      REAL ICN
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)

C---- INPUT AND PRINT WORKING VARIABLES

      READ *, FREQ1, AMP1, FS
      READ *,FC1, TC, RATIO
      READ *,BETA, GAMMA
      PRINT *, ' IDLE CHANNEL NOISE TEST AT ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC,' AND RATIO = ',RATIO
      PRINT *, ' OUTPUT FILTER PARAMETERS ARE: BETA = ',BETA
      PRINT *, '                               GAMMA = ',GAMMA

C---- GENERATE THE FILTER COEFFICIENTS AND CVSD SYSTEM PARAMETERS

      CALL FLTRGEN(BETA,GAMMA,NP,B)
      CALL UMAXOPT(UMAX,UMIN,FS,FC1,TC,RATIO)

C---- GENERATE INPUT TIME FUNCTION SAMPLES

      PEAK1 = A (AMP1)
      CALL SIGNAL(TSIN,5000,FS,FREQ1,0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME FUNCTION THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE

C---- CALCULATE THE REFERENCE SYSTEM GAIN

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN = SQRT (PIN/POUT)

C---- GENERATE A ZERO INPUT SIGNAL ARRAY

      DO 45 I = 1,5000
      TSIN(I) = 0.
45    CONTINUE

C---- PROCESS THE ZERO SIGNAL THROUGH THE SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,UMAX,UMIN,DC)

C---- FILTER THE OUTPUT SIGNAL

      CALL FILTER(TSOUT,5000,NP,B)

C---- ADJUST THE OUTPUT SIGNAL AMPLITUDE TO THE REFERENCE VALUE

      DO 46 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN
46    CONTINUE

C---- CALCULATE THE IDLE CHANNEL NOISE

      CALL POWER(TSOUT,4096,FS,POUT)
      ICN = 10. * ALOG10(POUT)

C---- PRINT OUT THE RESULTS

      WRITE(6,600) ICN
600   FORMAT(1X,'THE IDLE CHANNEL NOISE = ',F8.2)
      END
```

APPENDIX M

Total Harmonic Distortion Program

```
      PROGRAM HARDIST(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT)
C---------- ---------TOTAL HARMONIC DISTORTION PROGRAM--------------------

C   THIS PROGRAM CALCULATES THE TOTAL HARMONIC DISTORTION IN THE OUTPUT
C   WHEN A SINGLE FREQUENCY TEST SIGNAL IS PROCESSED THROUGH A CVSD
C   ENCODER AND DECODER CONNECTED BACK-TO-BACK.  DISTORTION IS CALCULATED
C   USING ONLY THOSE SPECTRAL COMPONENTS OF THE OUTPUT THAT LIE BETWEEN
C   100 HZ AND 4000 HZ.  THOSE AT EXACTLY 100 HZ OR 4000 HZ ARE NOT
C   INCLUDED IN THE CALCULATION.

C$$$$$$$$$$$$$$$$$$$$$$$ VARIABLES $$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$

C     INPUT = AN INTEGER ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD
C        ENCODER.

C     OUTPUT = A REAL ARRAY CONTAINING THE TIME FUNCTION OUTPUT OF THE
C        TEST SIGNAL GENERATOR AND AFTER PROCESSING, THE TIME FUNCTION
C        OUTPUT OF THE DECODER.

C     PSX = A REAL ARRAY CONTAINING THE OUTPUT SPECTRAL POWER COMPONENTS
C        OF THE DECODER OUTPUT AFTER PROCESSING BY THE FAST FOURIER
C        TRANSFORM SUBROUTINE.

C     IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.

C     FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C        HAS CALCULATED THE SPECTRAL COMPONENTS.

C     FREQ = THE FREQUENCY OF THE TEST SIGNAL IN HZ.

C     AMP = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.

C     FS = THE SAMPLE RATE IN BPS.

C     A = THE PEAK VALUE OF THE TEST SIGNAL.

C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES FOR THE PRINCIPLE INTEGRATOR
C        IN THE CVSD ENCODER AND DECODER.

C     TC = THE COMPANDING SPEED OF THE SYLLABIC INTEGRATOR IN SEC.

C     F = A FREQUENCY VARIABLE USED TO DETERMINE THE TEST SIGNAL SPEC-
C        TRAL COMPONENT AND ALSO TO DETERMINE THE HARMONIC SPECTRAL
C        COMPONENTS.

C     E0 = THE RMS POWER OF THE OUTPUT SPECTRAL COMPONENT AT THE TEST
C        FREQUENCY.

C     REF = THE OUTPUT SPECTRAL COMPONENT POWER IN DBM.

C     SUM = THE RUNNING SUM OF THE POWER OF THE HARMONIC COMPONENTS.

C     THD = THE TOTAL HARMONIC DISTORTION IN %.

C     BETA = NORMALIZED 3 DB FREQUENCY OF THE OUTPUT FILTER

C     GAMMA = THE NORMALIZED ROLL-OFF BANDWIDTH OF THE OUTPUT FILTER

C     RATIO = THE RATIO OF THE MAXIMUM STEP SIZE TO THE MINIMUM STEP
C        SIZE IN THE CVSD ENCODER AND DECODER, GIVEN ON DB.

C$$$$$$$$$$$$$$$$$$$$ SUBROUTINES USED $$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$

C     ENCODE1 = THE CVSD ENCODER

C     DECODE1 = THE CVSD DECODER

C     VMAXOPT = GENERATES VMAX AND VMIN USED IN THE CVSD ENCODER AND
C        DECODER

C     FLTERGEN = THE COEFFICENT GENERATOR FOR THE OUTPUT FILTER

C     FILTER = FILTERS THE OUTPUT SIGNAL.

C     FTFPS = THE FAST FOURIER TRANSFORM SUBROUTINE FROM THE IMSL
C        LIBRARY

C$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
C$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
```

```
C------------------PROGRAM START------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION INPUT(5000),OUTPUT(5000),PSX(150)
     1,IWK(20),WK(150),FREQUE(200), B(200)
      COMPLEX CWK(300)
      FREQ2 = 0.
      AMP2 = 0.

C---- INPUT AND PRINT THE WORKING VARIABLES

      READ *,  FREQ1, AMP1, FS
      READ *, FC1, TC, RATIO
      READ *, BETA, GAMMA
      PRINT *,' AMP = ',AMP1,' DBM0,  FREQ = ',FREQ1,' HZ,  SAMPLE RATE
     1= ',FS
      PRINT *,' TC = ',TC,',  FC1 = ',FC1,',  RATIO = ',RATIO
      PRINT *,' BETA = ',BETA,',  GAMMA = ',GAMMA

C---- DETERMINE PEAK VALUE OF TEST SIGNAL

      A = SQRT(10. ** ((AMP1 - 4.) / 10.) * .001 * 600.) * SQRT(2.)

C---- GENERATE INPUT TIME FUNCTION

      CALL SIGNAL(OUTPUT,5000,FS,FREQ1,FREQ2,A,AMP2)

C---- GENERATE THE FILTER COEFFICIENTS AND CVSD SYSTEM PARAMETERS

      CALL FLTRGEN(BETA,GAMMA,NP,B)
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)

C---- PROCESS THE TIME FUNCTION THROUGH THE CVSD SYSTEM

      CALL ENCODE1(OUTPUT,INPUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
      CALL DECODE1(INPUT,OUTPUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT SIGNAL

      CALL FILTER(OUTPUT,5000,NP,B)

C---- DETERMINE THE SPECTRAL COMPONENTS OF THE OUTPUT
C---- REMOVE THE MEAN OF THE SAMPLE STRING

      SUM = 0.
      DO 3 I = 1,4096
      SUM = SUM + OUTPUT(I)
3     CONTINUE
      AVER = SUM / 4096
      DO 4 I = 1,4096
      OUTPUT(I) = OUTPUT(I) - AVER
4     CONTINUE

      CALL FTFPS(OUTPUT,DUM,4096,256,0,PSX,DUM,DUM,IWK,WK,CWK,IER)

C---- DETERMINE THE COMPONENT AT THE TEST FREQUENCY AND CALCULATE
C     THE RMS VOLTAGE.

      KF = 0
      DO 8 K = 3,129,2
      KF = KF + 1
      F = ((K-1.)/256.) * FS
      FREQUE(KF) = F
      PSX(KF) = PSX(K)
      IF (F .NE. FREQ1) GO TO 8
      E0 = SQRT( PSX(K) )
      REF = 10. * ALOG10(PSX(K) / 600. /.001)
8     CONTINUE

C---- CALCULATE POWER AT HARMONIC FREQUENCIES

      SUM = 0.
      DO 20 I = 2,10
      F = FREQ1 / I
      IF ((F .LE. 100.) .OR. (F .GT. 4000.)) GO TO 20
      DO 15 J = 1,KF
      IF (F .NE. FREQUE(J)) GO TO 15
      SUM = SUM + PSX(J)
15    CONTINUE
20    CONTINUE
      DO 30 I = 2,30
      F = FREQ1 * I
      IF ((F .LE. 100.) .OR. (F .GE. 4000.)) GO TO 30
```

```
      DO 25 J = 1,KF
      IF ( F .NE. FREQUE(J)) GO TO 25
      SUM = SUM + PSX(J)
25    CONTINUE
30    CONTINUE

C---- CALCULATE TOTAL HARMONIC DISTORTION

      THD = SQRT (SUM) / E0 * 100.
      WRITE(6,606) THD
606   FORMAT(1X,'THE TOTAL HARMONIC DISTORTION IS ',F5.2,'%.')
      END
```

APPENDIX N

Total Harmonic Distortion vs. Input Signal Power

```
      PROGRAM DTHD(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT,PLOT)
C--------------------THD VS. INPUT POWER--------------------------
C     THIS PROGRAM INVESTIGATES THE VARIATION IN HARMONIC DISTORTION
C     AS THE SIGNAL INPUT POWER IS VARIED.  THE INPUT POWER IS CHANGED
C     IN .4 DB STEPS FROM -48 DBM0 TO 0 DBM0.  THE HARMONIC DISTORTION
C     IS THE OUTPUT IS THEN MEASURED AND PLOTTED.
C     THE PROGRAM IS REPEATED THREE TIMES, STEPPING THE STEP SIZE
C     RATIO FROM 32 DB TO 23 DB.  THE THREE SETS OF DATA ARE THEN
C     PLOTTED ON THE SAME GRAPH.
C********************** VARIABLES *******************************
C     INPUT = AN INTEGER ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD
C        ENCODER.
C     OUTPUT = A REAL ARRAY CONTAINING THE TIME FUNCTION OUTPUT OF THE
C        TEST SIGNAL GENERATOR AND AFTER PROCESSING, THE TIME FUNCTION
C        OUTPUT OF THE DECODER.
C     POWER = A REAL ARRAY CONTAINING THE POWER THAT EACH SAMPLE IS
C        TAKEN.
C     PSX = A REAL ARRAY CONTAINING THE OUTPUT SPECTRAL POWER COMPONENTS
C        OF THE DECODER OUTPUT AFTER PROCESSING BY THE FAST FOURIER
C        TRANSFORM SUBROUTINE.
C     IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.
C     FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C        HAS CALCULATED THE SPECTRAL COMPONENTS.
C     N = THE NUMBER OF TIME SAMPLES TO BE TAKEN.
C     FREQ = THE FREQUENCY OF THE TEST SIGNAL IN HZ.
C     AMP = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE IN BPS.
C     A = THE PEAK VALUE OF THE TEST SIGNAL.
C     IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.
C     FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C        HAS CALCULATED THE SPECTRAL COMPONENTS.
C     N = THE NUMBER OF TIME SAMPLES TO BE TAKEN.
C     FREQ = THE FREQUENCY OF THE TEST SIGNAL IN HZ.
C     AMP = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE IN BPS.
C     A = THE PEAK VALUE OF THE TEST SIGNAL.
C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES FOR THE PRINCIPLE INTEGRATOR
C        IN THE CVSD ENCODER AND DECODER.
C     TC = THE COMPANDING SPEED OF THE SYLLABIC INTEGRATOR IN SEC.
C     I, J, K = COUNTING INDICES FOR THE VARIOUS 'DO' LOOPS.
C     F = A FREQUENCY VARIABLE USED TO DETERMINE THE TEST SIGNAL SPEC-
C        TRAL COMPONENT AND ALSO TO DETERMINE THE HARMONIC SPECTRAL
C        COMPONENTS.
C     E0 = THE RMS POWER OF THE OUTPUT SPECTRAL COMPONENT AT THE TEST
C        FREQUENCY.
C     REF = THE OUTPUT SPECTRAL COMPONENT POWER IN DBM.
```

```
C     SUM = THE RUNNING SUM OF THE POWER OF THE HARMONIC COMPONENTS.
C     THD = AN ARRAY CONTAINING THE VALUE OF HARMONIC DISTORTION AT EACH
C        LEVEL OF INPUT POWER.
C     B = AN ARRAY CONTAINING THE OUTPUT FILTER COEFFICIENTS.
C     NP = THE NUMBER OF OUTPUT FILTER COEFFICIENTS.
C     BETA = THE NORMALIZED CENTER OF THE TRANSITION BAND FOR THE OUTPUT
C        FILTER.
C     GAMMA = THE NORMALIZED WDTH OF THE OUTPUT FILTER TRANSITION BAND.
C*************************************************************************
C*************************************************************************
C------------------PROGRAM START-------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION INPUT(5000),OUTPUT(5000),POWER(202),PSX(150)
     1,IWK(20),WK(150),FREQUE(150),  THD(202), B(200)
      COMPLEX CWK(300)
      A(DBM0) = SQRT(10. ** ((DBM0 - 4.) / 10.) * .001 * 600.) * SQRT
     1)
      ICHAR = -1
C---- INPUT WORKING VARIABLES
      READ *,  FREQ1,  FS
      PRINT *,' DYNAMIC RANGE TEST AT ',FS,' BPS AND ',FREQ1,' HZ'
      READ *, FC1, TC
      READ *, XLEN, YLEN, XMIN, XMAX, YMIN, YMAX
      XSTEP = (XMAX - XMIN) / XLEN
      YSTEP = (YMAX - YMIN) / YLEN
      PRINT *,' TC = ',TC,', FC1 = ',FC1
      READ *,BETA, GAMMA
      PRINT *,' BETA = ',BETA,', GAMMA = ',GAMMA
      CALL FLTRGEN(BETA,GAMMA,NP,B)
C---- START LOOP
      DO 1000 NR = 2,6,2
      RATIO = 30. + NR
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      DO 500 IS = 1,100
      POWER(IS) = -40. + .4 * IS
C---- DETERMINE PEAK VALUE OF TEST SIGNAL
      AMP1 = A(POWER(IS))
C---- GENERATE INPUT TIME FUNCTION
      CALL SIGNAL(OUTPUT,5000,FS,FREQ1,0.,AMP1,0.)
C---- PROCESS THE TIME FUNCTION THROUGH THE CVSD SYSTEM
      CALL ENCODE1(OUTPUT,INPUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
      CALL DECODE1(INPUT,OUTPUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
C---- FILTER THE OUTPUT
      CALL FILTER(OUTPUT,5000,NP,B)
C---  DETERMINE THE SPECTRAL COMPONENTS OF THE OUTPUT
      CALL FTFPS(OUTPUT,DUM,4096,256,0,PSX,DUM,DUM,IWK,WK,CWK,IER)
C---- DETERMINE THE COMPONENT AT THE TEST FREQUENCY AND
C     CALCULATE REFERENCE VALUES.
```

```
      KF = 0
      DO 8 K = 3,129,2
      KF = KF + 1
      F = ((K-1)/256.) * FS
      FREQUE(KF) = F
      IF (PSX(K) .LE. 6.E-19) PSX(K) = 6.E-10
      PSX(KF) = PSX(K)
      IF (F .NE. FREQ1) GO TO 8
      E0 = SQRT( PSX(K) )
8     CONTINUE

C---- CALCULATE POWER AT HARMONIC FREQUENCIES

      SUM = 0.
      DO 20 I = 2,10
      F = FREQ1 / I
      IF ((F .LE. 100.) .OR. (F .GT. 4000.)) GO TO 20
      DO 15 J = 1,KF
      IF (F .NE. FREQUE(J)) GO TO 15
      SUM = SUM + PSX(J)
15    CONTINUE
20    CONTINUE
      DO 30 I = 2,30
      F = FREQ1 * I
      IF ((F .LE. 100.) .OR. (F .GE. 4000.)) GO TO 30
      DO 25 J = 1,KF
      IF ( F .NE. FREQUE(J)) GO TO 25
      SUM = SUM + PSX(J)
25    CONTINUE
30    CONTINUE

C---- CALCULATE TOTAL HARMONIC DISTORTION

      THD(IS) = SQRT (SUM) / E0 * 100.
      IF (THD(IS) .GT. 100.) THD(IS) = 100.
500   CONTINUE

C---- PLOT RESULTS

      ICHAR = ICHAR + 1
      CALL PLTRANG(POWER,THD,100,ICHAR,XMIN,XLEN,XSTEP,YMIN,YLEN,YSTEP)
1000  CONTINUE
      CALL PLOTE(N)
      END

      SUBROUTINE PLTRANG(X,Y,N,ICHAR,XMIN,XLEN,XSTEP,YMIN,YLEN,YSTEP)

C---------------DYNAMIC RANGE PLOT SUBROUTINE-----------------------

C     THIS SUBROUTINE CREATES A PLOT OF THE Y ARRAY VERSUS THE X ARRAY.

C************************* VARIABLES ********************************

C     X = AN ARRAY CONTAINING THE ORDINATE VALUES

C     Y = AN ARRAY CONTAINING THE VALUES TO BE PLOTTED.

C     N = THE NUMBER OF VALUES IN THE ARRAYS

C******************************************************************
C******************************************************************

C---------------------SUBROUTINE START-----------------------------

C---- INITIALIZE ARRAYS

      DIMENSION X(4096), Y(4096)
      X(N+1) = XMIN
      X(N+2) = XSTEP
      Y(N+1) = YMIN
      Y(N+2) = YSTEP
      IF (ICHAR .GT. 0) GO TO 500

C---- ESTABLISH NEW PAGE ORIGIN

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)
```

```
C---- BOX IN THE GRAPH

      CALL RECT(0., 0., YLEN, XLEN, 0., 3)

C---- DRAW THE AXES

      CALL AXIS(0.0,0.0,18HINPUT POWER (DBM0),-18,XLEN,0.0,XMIN,XSTEP)
      CALL AXIS(0.0,0.0,14HDISTORTION (%),14,YLEN,90.0,YMIN,YSTEP)

C---- PLOT VALUES

500   CONTINUE
      CALL LINE(X,Y,N,1,10,ICHAR)
      RETURN
      END
```

APPENDIX O

Mismatched Total Harmonic Distortion vs. Input Signal Power

```
      PROGRAM MMTHD(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT,PLOT)
C------------MISMATCHED THD VS. INPUT POWER--------------------------
C     THIS PROGRAM INVESTIGATES THE VARIATION IN HARMONIC DISTORTION
C     AS THE SIGNAL INPUT POWER IS VARIED.  THE INPUT POWER IS CHANGED
C     IN .4 DB STEPS FROM -40 DBM0 TO 0 DBM0.  THE HARMONIC DISTORTION
C     IS THE OUTPUT IS THEN MEASURED AND PLOTTED.
C     THE PROGRAM IS REPEATED THREE TIMES.  WHILE THE ENCODER PARAMETERS
C     ARE HELD CONSTANT, THE DECODER STEP SIZE RATIO IS ALLOWED TO VARY
C     FROM 32 DB TO 33 DB.  THE THREE SETS OF DATA ARE THEN PLOTTED ON
C     THE SAME GRAPH.
C********************** VARIABLES *************************************
C     INPUT = AN INTEGER ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD
C        ENCODER.
C     OUTPUT = A REAL ARRAY CONTAINING THE TIME FUNCTION OUTPUT OF THE
C        TEST SIGNAL GENERATOR AND AFTER PROCESSING, THE TIME FUNCTION
C        OUTPUT OF THE DECODER.
C     POWER = A REAL ARRAY CONTAINING THE POWER THAT EACH SAMPLE IS
C        TAKEN.
C     PSX = A REAL ARRAY CONTAINING THE OUTPUT SPECTRAL POWER COMPONENTS
C        OF THE DECODER OUTPUT AFTER PROCESSING BY THE FAST FOURIER
C        TRANSFORM SUBROUTINE.
C     IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.
C     FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C        HAS CALCULATED THE SPECTRAL COMPONENTS.
C     N = THE NUMBER OF TIME SAMPLES TO BE TAKEN.
C     FREQ = THE FREQUENCY OF THE TEST SIGNAL IN HZ.
C     AMP = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE IN BPS.
C     A = THE PEAK VALUE OF THE TEST SIGNAL.
C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES FOR THE PRINCIPLE INTEGRATOR
C        IN THE CVSD ENCODER AND DECODER.
C     TC = THE COMPANDING SPEED OF THE SYLLABIC INTEGRATOR IN SEC.
C     I, J, K = COUNTING INDICES FOR THE VARIOUS 'DO' LOOPS.
C     F = A FREQUENCY VARIABLE USED TO DETERMINE THE TEST SIGNAL SPEC-
C        TRAL COMPONENT AND ALSO TO DETERMINE THE HARMONIC SPECTRAL
C        COMPONENTS.
C     E0 = THE RMS POWER OF THE OUTPUT SPECTRAL COMPONENT AT THE TEST
C        FREQUENCY.
C     REF = THE OUTPUT SPECTRAL COMPONENT POWER IN DBM.
C     SUM = THE RUNNING SUM OF THE POWER OF THE HARMONIC COMPONENTS.
C     THD = AN ARRAY CONTAINING THE VALUE OF HARMONIC DISTORTION AT EACH
C        LEVEL OF INPUT POWER.
C*********************************************************************
C*********************************************************************
```

```
C-----------------PROGRAM START--------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION INPUT(5000),OUTPUT(5000),POWER(202),PSX(150)
     1,IWK(20),WK(150),FREQUE(150), THDREF(202), THD(202), B(200)
      COMPLEX CWK(300)
      A(DBM0) = SQRT(10. ** ((DBM0 - 4.) / 10.) * .001 * 600.) * SQRT(2.
     1)
      ICHAR = -1

C---- INPUT WORKING VARIABLES

      READ *,  FREQ1,  FS
      PRINT *,' DYNAMIC RANGE TEST AT ',FS,' BPS AND ',FREQ1,' HZ'
      READ *, FC1, TC
      READ *, XLEN, YLEN, XMIN, XMAX, YMIN, YMAX
      XSTEP = (XMAX - XMIN) / XLEN
      YSTEP = (YMAX - YMIN) / YLEN
      PRINT *,' TC = ',TC,', FC1 = ',FC1
      READ *,BETA, GAMMA
      PRINT *,' BETA = ',BETA,', GAMMA = ',GAMMA
      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- START LOOP

      DO 1000 NR = 2,6,2
      RATIO = 33. + NR
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      IF (ICHAR .GE. 0) GO TO 2
      EVMX = VMAX
      EVMN = VMIN
2     CONTINUE
      DO 500 IS = 1,100
      POWER(IS) = -40. + .4 * IS

C---- DETERMINE PEAK VALUE OF TEST SIGNAL

      AMP1 = A(POWER(IS))

C---- GENERATE INPUT TIME FUNCTION

      CALL SIGNAL(OUTPUT,5000,FS,FREQ1,0.,AMP1,0.)

C---- PROCESS THE TIME FUNCTION THROUGH THE CVSD SYSTEM

      CALL ENCODE1(OUTPUT,INPUT,5000,FS,FC1,FC2,FC3,TC,EVMX,EVMN,DC)
      CALL DECODE1(INPUT,OUTPUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT

      CALL FILTER(OUTPUT,5000,NP,B)

C---  DETERMINE THE SPECTRAL COMPONENTS OF THE OUTPUT

      CALL FTFPS(OUTPUT,DUM,4096,256,0,PSX,DUM,DUM,IWK,WK,CWK,IER)

C---- ELIMINATE THE COMPONENTS AT ODD MULTIPLES OF THE SAMPLE RATE.
C     DETERMINE THE COMPONENT AT THE TEST FREQUENCY AND
C     CALCULATE REFERENCE VALUES.

      KF = 0
      DO 8 K = 3,129,2
      KF = KF + 1
      F = ((K-1)/256.) * FS
      FREQUE(KF) = F
      IF (PSX(K) .LE. 6.E-10) PSX(K) = 6.E-10
      PSX(KF) = PSX(K)
      IF (F .NE. FREQ1) GO TO 8
      E0 = SQRT( PSX(K) )
8     CONTINUE

C---- CALCULATE POWER AT HARMONIC FREQUENCIES
```

```
      SUM = 0.
      DO 20 I = 2,10
      F = FREQ1 / I
      IF ((F .LE. 100.) .OR. (F .GT. 4000.)) GO TO 20
      DO 15 J = 1,KF
      IF (F .NE. FREQUE(J)) GO TO 15
      SUM = SUM + PSX(J)
15    CONTINUE
20    CONTINUE
      DO 30 I = 2,30
      F = FREQ1 * I
      IF ((F .LE. 100.) .OR. (F .GE. 4000.)) GO TO 30
      DO 25 J = 1,KF
      IF ( F .NE. FREQUE(J)) GO TO 25
      SUM = SUM + PSX(J)
25    CONTINUE
30    CONTINUE
C---- CALCULATE TOTAL HARMONIC DISTORTION
      THD(IS) = SQRT (SUM) / E0 * 100.
      IF (THD(IS) .GT. 100.) THD(IS) = 100.
500   CONTINUE
C---- PLOT RESULTS
      ICHAR = ICHAR + 1
      CALL PLTRANG(POWER,THD,100,ICHAR,XMIN,XLEN,XSTEP,YMIN,YLEN,YSTEP)
1000  CONTINUE
      CALL PLOTE(N)
      END
C*********************************************************************** ***
C*********************************************************************** ***
      SUBROUTINE PLTRANG(X,Y,N,ICHAR,XMIN,XLEN,XSTEP,YMIN,YLEN,YSTEP)
C---------------DYNAMIC RANGE PLOT SUBROUTINE----------------------------
C     THIS SUBROUTINE CREATES A SEMI-LOG PLOT OF THE X AND Y ARRAYS.
C************************ VARIABLES *************************************
C     X = AN ARRAY CONTAINING THE ORDINATE VALUES
C     Y = AN ARRAY CONTAINING THE VALUES TO BE PLOTTED.
C     N = THE NUMBER OF VALUES IN THE ARRAYS
C************************************************************************
C************************************************************************
C---------------------SUBROUTINE START-----------------------------------
C---- INITIALIZE ARRAYS
      DIMENSION X(4096), Y(4096)
      X(N+1) = XMIN
      X(N+2) = XSTEP
      Y(N+1) = YMIN
      Y(N+2) = YSTEP
      IF (ICHAR .GT. 0) GO TO 500
C---- ESTABLISH NEW PAGE ORIGIN
      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)
C---- BOX IN THE GRAPH
      CALL RECT(0., 0., YLEN, XLEN, 0., 3)
C---- DRAW THE AXES
      CALL AXIS(0.0,0.0,18HINPUT POWER (DBM0),-18,XLEN,0.0,XMIN,XSTEP)
      CALL AXIS(0.0,0.0,14HDISTORTION (%),14,YLEN,90.0,YMIN,YSTEP)
```

```
C---- PLOT VALUES

500   CONTINUE
      CALL LINE(X,Y,N,1,10,ICHAR)
      RETURN
      END
```

APPENDIX P

Intermodulation Distortion Program

```
      PROGRAM INTERMD(INPUT,OUTPUT,TAPE5=INPUT,TAPE6=OUTPUT)
C---------------INTERMODULATION DISTORTION PROGRAM--------------------
C      THIS PROGRAM CALCULATES THE INTERMODULATION DISTORTION OF A
C      CVSD SYSTEM WHERE THE ENCODER AND DECODER ARE CONNECTED BACK-TO-
C      BACK.  DISTORTION IS MEASURED BY INPUTTING A TEST SIGNAL COMPOSED
C      OF TWO EQUAL AMPLITUDE SINUSOIDS AT [illegible] HZ AND [illegible] HZ.  THE
C      AMPLITUDE OF THE DIFFERENCE PRODUCT IS THEN MEASURED AND COMPARED
C      TO THE INPUT SIGNAL TO DETERMINE THE PERCENT DISTORTION.
C********************** VARIABLES ************************************
C      INPUT = AN INTEGER ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD
C         ENCODER.
C      OUTPUT = A REAL ARRAY CONTAINING THE TIME FUNCTION OUTPUT OF THE
C         TEST SIGNAL GENERATOR AND AFTER PROCESSING, THE TIME FUNCTION
C         OUTPUT OF THE DECODER.
C      PSX = A REAL ARRAY CONTAINING THE OUTPUT SPECTRAL POWER COMPONENTS
C         OF THE DECODER OUTPUT AFTER PROCESSING BY THE FAST FOURIER
C         TRANSFORM (FFT) SUBROUTINE.
C      IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.
C      FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C         HAS CALCULATED THE SPECTRAL COMPONENTS.
C      N = THE NUMBER OF TIME SAMPLES TO BE TAKEN.
C      AMP1 = THE AMPLITUDES OF THE TEST SIGNALS IN DBM0.
C      FS = THE SAMPLE RATE IN BPS.
C      PEAK = THE PEAK VALUES OF THE TEST SIGNAL COMPONENTS.
C      A 'D' OR 'E' PREFIX ON THE NEXT THREE SETS OF VARIABLE INDICATES
C         THE VARIABLE IS USED BY EITHER THE DECODER OR ENCODER,RESPEC-
C         TIVELY
C      FC1, FC2, FC3 = ROLL-OFF FREQUENCIES FOR THE PRINCIPLE INTEGRATOR
C         IN THE CVSD ENCODER AND DECODER.
C      TC = THE COMPANDING SPEED OF THE SYLLABIC INTEGRATOR IN SEC.
C      RATIO = THE MAXIMUM STEP SIZE TO MINIMUM STEP SIZE RATIO IN DB
C      F = A FREQUENCY VARIABLE USED TO DETERMINE THE TEST SIGNAL SPEC-
C         TRAL COMPONENT AND ALSO TO DETERMINE THE HARMONIC SPECTRAL
C         COMPONENTS.
C      REF = THE OUTPUT SPECTRAL COMPONENT POWER IN DBM.
C      SUM = THE SUM OF THE POWER IN THE TEST SIGNAL COMPONENTS.
C      NP = THE NUMBER OF FILTER COEFFICIENTS
C      B = AN ARRAY CONTAINING THE INPUT AND OUTPUT FILTER COEFFICIENTS
*********************** SUBROUTINES USED ****************************
C      ENCODE1 = THE CVSD ENCODER
C      DECODE1 = THE CVSD DECODER
C      VMAXOPT = DETERMINES THE VALUES OF VMAX AND VMIN USED IN THE
C         ENCODER AND DECODER
C      FLTRGEN = GENERATOR OF THE OUTPUT FILTER COEFFICIENTS
C      FILTER = FILTERS THE OUTPUT USING THE COEFFICIENTS PRODUCED BY
C         FLTRGEN
C      SIGNAL = THE TEST SIGNAL GENERATOR
C      FTFPS = THE FAST FOURIER TRANSFORM FROM THE IMSL LIBRARY
C*********************************************************************
```

```
C------------------PROGRAM START------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION INPUT(5000),OUTPUT(5000),PSX(150)
     1,IUK(20),WK(150),FREQUE(200),  B(200)
      REAL IMOD
      COMPLEX CWK(300)
      A(DBM0) = SQRT (10. ** ((DBM0 - 4.)/10.) * .001 * 600.) * SQRT(2.)

C---- INPUT AND PRINT WORKING VARIABLES

      READ *, AMP1, FS
      READ *, EFC1, ETC, ERATIO
      READ *, DFC1, DTC, DRATIO
      READ *, BETA, GAMMA
      PRINT *,' INTERMOD TEST FOR INPUT SIGNALS OF 750 AND 1000 HZ AT ',
     1AMP1,' DBM0, AND SAMPLE RATE = ',FS,' K3/S'
      PRINT *,' ETC = ',ETC,', EFC1 = ',EFC1,', ERATIO = ',ERATIO
      PRINT *,' DTC = ',DTC,', DFC1 = ',DFC1,', DRATIO = ',DRATIO
      PRINT *,' BETA = ',BETA,', GAMMA = ',GAMMA

C---- DETERMINE PEAK VALUE OF TEST SIGNAL

      PEAK = A (AMP1)

C---- GENERATE INPUT TIME FUNCTION

      CALL SIGNAL(OUTPUT, 5000, FS, 750., 1000., PEAK, PEAK)

C---- GENERATE FILTER COEFFICIENTS AND CVSD SYSTEM PARAMETERS

      CALL VMAXOPT(EVMX,EVMN,FS,EFC1,ETC,ERATIO)
      CALL VMAXOPT(DVMX,DVMN,FS,DFC1,DTC,DRATIO)
      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- PROCESS THE TIME FUNCTION THROUGH THE CVSD SYSTEM

      CALL ENCODE1(OUTPUT,INPUT,5000,FS,EFC1,FC2,FC3,ETC,EVMX,EVMN,DC)
      CALL DECODE1(INPUT,OUTPUT,5000,FS,DFC1,FC2,FC3,DTC,DVMX,DVMN,DC)

C---- FILTER THE OUTPUT

      CALL FILTER(OUTPUT,5000,NP,B)

C---- SUBTRACT THE AVERAGE VALUE FROM THE SAMPLE STRING

      SUM = 0.
      DO 6 I = 1,4096
      SUM = SUM + OUTPUT(I)
6     CONTINUE
      AVER = SUM / 4096
      DO 7 I = 1,4096
      OUTPUT(I) = OUTPUT(I) - AVER
7     CONTINUE

C---- CALCULATE THE SPECTRAL COMPONENTS OF THE OUTPUT

      CALL FTFPS(OUTPUT,DUM,4096,256,0,PSX,DUM,DUM,IWK,WK,CWK,IER)

C     DETERMINE THE COMPONENT AT THE TEST FREQUENCY AND CALCULATE
C     THE RMS VOLTAGE.

      SUM = 0.
      DO 8 K = 2,129
      F = ((K-1)/256.) * FS
      FREQUE(K-1) = F
      PSX(K-1) = PSX(K)
      IF ((F .NE. 750.) .AND. (F .NE. 1000.)) GO TO 8
      SUM = SUM + SQRT ( PSX(K) )
      REF = 10. * ALOG10(PSX(K) / 600. /.001)
8     CONTINUE

C---- CALCULATE INTERMODULATION DISTORTION

      DIF = 250.
      DO 15 I = 1,128
      IF (FREQUE(I) .NE. DIF) GO TO 15
      DIS = PSX(I)
15    CONTINUE
      IMOD = SQRT (DIS) / SUM * 100.
      IF (IMOD .GT. 100.) IMOD = 100.
      WRITE(6,606) IMOD
606   FORMAT(1X,'THE INTERMODULATION DISTORTION IS ',F5.2,'%.')

      END
```

APPENDIX Q

Intermodulation Distortion vs. Input Signal Power

```
      PROGRAM DIMOD(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C-----------------------INTERMOD VS. INPUT POWER--------------------
C     THIS PROGRAM MEASURES INTERMODULATION DISTORTION AS A FUNCTION OF
C     INPUT SIGNAL POWER IN A CVSD SYSTEM WHERE THE ENCODER AND DECODER
C     PARAMETERS ARE PERFECTLY MATCHED.  CALCULATIONS ARE REPEATED THREE
C     TIMES AS THE STEP SIZE RATIO IS VARIED FROM 33 DB TO 23 DB.
C     THE THREE SETS OF DATA ARE THEN PLOTTED ON THE SAME GRAPH.
C*********************** VARIABLES **********************************
C     X = AN ARRAY CONTAINING THE TEST SIGNAL INPUT POWER AT WHICH MEA-
C         SUREMENTS ARE MADE.
C     Y = AN ARRAY CONTAINING THE INTERMODULATION DISTORTION MEASURE-
C         MENTS.
C     B = AN ARRAY CONTAINING THE OUTPUT FILTER COEFFICIENTS.
C     IMOD = INTERMODULATION DISTORTION AT THE PRESENT TEST SIGNAL POWER
C     FS = THE SAMPLE RATE
C**   THE FOLLOWING VARIABLES USE AN 'E' AND A 'D' PREFIX TO INDICATE
C     USE BY EITHER THE ENCODER OR DECODER, RESPECTIVELY.
C        FC1 = ROLL-OFF FREQUENCY OF THE PRIMARY INTEGRATOR
C        TC = THE TIME CONSTANT OF THE SYLLABIC FILTER
C        VMX = THE MAXIMUM INPUT TO THE SYLLABIC FILTER
C        VMN = THE MINIMUM INPUT TO THE SYLLABIC FILTER
C     BETA = THE NORMALIZED CENTER FREQUENCY OF THE OUTPUT FILTER TRANS-
C         ITION BAND
C     GAMMA = THE NORMALIZED WIDTH IF THE OUTPUT FILTER TRANSITION BAND
C*********************** SUBROUTINES USED ***************************
C     FLTRGEN = THE OUTPUT FILTER COEFFICIENT GENERATOR
C     FILTER = THE SUBROUTINE THAT FILTERS THE DECODER OUTPUT SIGNAL
C        USING THE COEFFICIENTS CALCULATED BY FLTRGEN.
C     VMAXOPT = CALCULATES VMAX AND VMIN FOR THE CVSD ENCODER
C        AND DECODER SUBROUTINES.
C     INTERMD = THE SUBROUTINE THAT CALCULATES THE INTERMODULATION DIS-
C        TORTION IN THE CVSD SYSTEM OUTPUT SIGNAL.
C     FACTOR, PLOT, SCALE, AXIS, RECT, LINE = CALCOMP PLOTTER ROUTINES
C*******************************************************************
C----------------------PROGRAM START--------------------------------
C---- INITIALIZE VARAIBLES AND ARRAYS
      DIMENSION X(150), Y(150), B(200)
      REAL IMOD
      ICHAR = -1
C---- READ AND PRINT THE WORKING VARIABLES
      READ *, FS
      READ *, EFC1, ETC
      READ *, DFC1, DTC
      READ *, BETA, GAMMA
      PRINT *,' FS = ',FS
      PRINT *,' EFC1 = ',EFC1,', ETC = ',ETC
      PRINT *,' DFC1 = ',DFC1,', DTC = ',DTC
      PRINT *,' BETA = ',BETA,', GAMMA = ',GAMMA
C---- GENERATE LOW-PASS FILTER COEFFICIENTS
      CALL FLTRGEN(BETA,GAMMA,NP,B)
C---- START CALCULATION LOOP
      DO 2000 J = 2,6,2
```

```
C---- CALCULATE CVSD SYSTEM PARAMETERS

      ICHAR = ICHAR + 1
      ERATIO = 39. + J
      CALL VMAXOPT(EVMX,EVMN,FS,EFC1,ETC,ERATIO)
100   CONTINUE
      DRATIO = 39. + J
      CALL VMAXOPT(DVMX,DVMN,FS,DFC1,DTC,DRATIO)

C---- CALCULATE INTERMODULATION DISTORTION VS. INPUT POWER

      DO 1000 I = 1,100
      AMP1 = -40. + .4 * I
      X(I) = AMP1
      CALL INTERMD(FS,AMP1,EFC1,ETC,EVMX,EVMN,DFC1,DTC,DVMX,DVMN,NP,B,IM
     1OD)
      Y(I) = IMOD
1000  CONTINUE

C---- PLOT THE RESULTS

      IF (ICHAR .GT. 0) GO TO 900
      CALL FACTOR(.5)
      CALL PLOT(2.,2.,-3)
      CALL SCALE(X,10.,100,1)
      CALL SCALE(Y,6.,100,1)
      CALL AXIS(0.,0.,1[illegible]DISTORTION (%),14,6.,90.,Y(101),Y(102))
      CALL AXIS(0.,0.,1[illegible]PLITUDE (DBM0),-16,10.,0.,X(101),X(102))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LINE(X,Y,100,1,10,ICHAR)
2000  CONTINUE
      CALL PLOTE(N)
      END
      SUBROUTINE INTERMD(FS,AMP1,EFC1,ETC,EVMX,EVMN,DFC1,DTC,DVMX,DVMN,N
     1P,B,IMOD)

C--------------INTERMODULATION DISTORTION SUBROUTINE------------------

C     THIS PROGRAM CALCULATES THE INTERMODULATION DISTORTION OF A
C     CVSD SYSTEM WHERE THE ENCODER AND DECODER ARE CONNECTED BACK-TO-
C     BACK.  DISTORTION IS [illegible] BY INPUTTING A TEST SIGNAL COMPOSED
C     OF TWO EQUAL AMPLITUDE [illegible] AT [illegible] HZ [illegible] HZ.  THE
C     AMPLITUDE OF THE DIFFERENCE PRODUCT IS THEN [illegible] AND COMPARED
C     TO THE INPUT SIGNAL TO DETERMINE THE PERCENT DISTORTION.

C*********************** VARIABLES *************************************


C     INPUT = AN INTEGER ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD
C        ENCODER.

C     OUTPUT = A REAL ARRAY CONTAINING THE TIME FUNCTION OUTPUT OF THE
C        TEST SIGNAL [illegible] AND AFTER PROCESSING, THE TIME FUNCTION
C        OUTPUT OF THE DECODER.

C     PSX = A REAL ARRAY CONTAINING THE OUTPUT SPECTRAL POWER COMPONENTS
C        OF THE DECODER OUTPUT AFTER PROCESSING BY THE FAST FOURIER
C        TRANSFORM (FFT) SUBROUTINE.

C     IWK, WK, CWK = WORKING ARRAYS USED BY THE FFT SUBROUTINE.

C     FREQUE = A REAL ARRAY CONTAINING THE FREQUENCIES AT WHICH THE FFT
C        HAS CALCULATED THE SPECTRAL COMPONENTS.

C     AMP1 = THE AMPLITUDES OF THE TEST SIGNALS IN DBM0.

C     FS = THE SAMPLE RATE IN BPS.

C     PEAK = THE PEAK VALUES OF THE TEST SIGNAL COMPONENTS.

C     A 'D' OR 'E' PREFIX ON THE NEXT THREE SETS OF VARIABLE INDICATES
C        THE VARIABLE IS USED BY EITHER THE DECODER OR ENCODER,RESPEC-
C        TIVELY

C     FC1, FC2, FC3 = ROLL-OFF FREQUENCIES FOR THE PRINCIPLE INTEGRATOR
C        IN THE CVSD ENCODER AND DECODER.

C     TC = THE COMPANDING SPEED OF THE SYLLABIC INTEGRATOR IN SEC.

C     RATIO = THE MAXIMUM STEP SIZE TO MINIMUM STEP SIZE RATIO IN DB

C     F = A FREQUENCY VARIABLE USED TO DETERMINE THE TEST SIGNAL SPEC-
C        TRAL COMPONENT AND ALSO TO DETERMINE THE HARMONIC SPECTRAL
C        COMPONENTS.

C     REF = THE OUTPUT SPECTRAL COMPONENT POWER IN DBM.

C     SUM = THE SUM OF THE POWER IN THE TEST SIGNAL COMPONENTS.
```

```
C     NP = THE NUMBER OF FILTER COEFFICIENTS
C     B = AN ARRAY CONTAINING THE INPUT AND OUTPUT FILTER COEFFICIENTS
********************** SUBROUTINES USED ******************************
C     ENCODE1 = THE CVSD ENCODER
C     DECODE1 = THE CVSD DECODER
C     FLTRGEN = GENERATOR OF THE OUTPUT FILTER COEFFICIENTS
C     FILTER = FILTERS THE OUTPUT USING THE COEFFICIENTS PRODUCED BY
C        FLTRGEN
C     SIGNAL = THE TEST SIGNAL GENERATOR
C     FTFPS = THE FAST FOURIER TRANSFORM FROM THE IMSL LIBRARY
C*********************************************************************
C-------------------SUBROUTINE START-----------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION INPUT(5000),OUTPUT(5040),PSX(150)
     1,IUK(20),UK(150),FREQUE(200),  B(200)
      REAL IMOD
      COMPLEX CWK(200)
      A(DBM0) = SQRT (10. ** ((DBM0 - 4.)/10.) * .001 * 600.) * SQRT(2.)
C---- DETERMINE PEAK VALUE OF TEST SIGNAL
      PEAK = A (AMP1)
C---- GENERATE INPUT TIME FUNCTION
      CALL SIGNAL(OUTPUT, 5000, FS, 750., 1000., PEAK, PEAK)
C---- PROCESS THE TIME FUNCTION THROUGH THE CVSD SYSTEM
      CALL ENCODE1(OUTPUT,INPUT,5000,FS,EFC1,FC3,FC3,ETC,EVMX,EVMN,DC)
      CALL DECODE1(INPUT,OUTPUT,5000,FS,DFC1,FC2,FC3,DTC,DVMX,DVMN,DC)
C---- FILTER THE OUTPUT
      CALL FILTER(OUTPUT,5000,NP,B'
C---- SUBTRACT THE AVERAGE VALUE FROM THE SAMPLE STRING
      SUM = 0.
      DO 6 I = 1,4096
      SUM = SUM + OUTPUT(I)
6     CONTINUE
      AVER = SUM / 4096
      DO 7 I = 1,4096
      OUTPUT(I) = OUTPUT(I) - AVER
7     CONTINUE
C---- CALCULATE THE SPECTRAL COMPONENTS OF THE OUTPUT
      CALL FTFPS(OUTPUT,DUM,4096,256,0,PSX,DUM,DUM,IUK,UK,CWK,IER)
C     DETERMINE THE COMPONENT AT THE TEST FREQUENCY AND CALCULATE
C     THE RMS VOLTAGE.
      SUM = 0.
      DO 8 K = 2,129
      F = ((K-1)/256.) * FS
      FREQUE(K-1) = F
      PSX(K-1) = PSX(K)
      IF ((F .NE. 750.) .AND. (F .NE. 1000.)) GO TO 8
      SUM = SUM + SQRT ( PSX(K) )
      REF = 10. * ALOG10(PSX(K) / 600. /.001)
8     CONTINUE
C---- CALCULATE INTERMODULATION DISTORTION
      DIF = 250.
      DO 15 I = 1,128
      IF (FREQUE(I) .NE. DIF) GO TO 15
      DIS = PSX(I)
15    CONTINUE
      IMOD = SQRT (DIS) / SUM * 100.
      IF (IMOD .GT. 100.) IMOD = 100.
      RETURN
      END
```

APPENDIX R

Signal-to-Noise Ratio Program

```
      PROGRAM SNRRAN(INPUT,OUTPUT,TAPE6=OUTPUT)
C---------------------FILTERRED SNR PROGRAM----------------------
C
C     THIS PROGRAM MEASURES THE SIGNAL-TO-NOISE PERFORMANCE OF A CVSD
C     ENCODER AND DECODER CONNECTED BACK-TO-BACK.  A SINGLE FREQUENCY
C     SINE WAVE IS INPUT TO THE SYSTEM AND THE DIFFERENCE BETWEEN THE
C     OUTPUT AND INPUT COMPUTED.
C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3600 HZ.
C
C$$$$$$$$$$$$$$$$$$$$$$$$ VARIABLES $$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
C
C     FREQ1 = THE TEST SIGNAL FREQUENCY
C
C     SNR = THE MEASURED SNR VALUE
C
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME SERIES SAMPLES.
C
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME SERIES
C        SAMPLES, THEN THE OUTPUT TIME SERIES SAMPLES OF THE FIR FILTER.
C
C     ERR = AN ARRAY CONTAINING THE DIFFERENCE BETWEEN THE OUTPUT SAM-
C        PLES AND THE INPUT SAMPLES.
C
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C
C     FS = THE SAMPLE RATE.
C
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C
C     BETA = THE NORMALIZED 3 DB FREQUENCY OF THE OUTPUT FILTER.  THE
C        FREQUENCY IS NORMALIZED TO THE SAMPLE RATE.
C
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C
C$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
C$$$$$$$$$$$$$$$$$$$$ SUBROUTINES USED $$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
C
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
C
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.
C
C     DECODE1 = THE CVSD DECODING SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.
C
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME SERIES SAMPLES
C        USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME SERIES
C        WITH IMPEDENCE = 600 OHMS.
C
C$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
C$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$$
```

```
C------------------PROGRAM START-----------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION   TSIN(5000), TSOUT(5000), ERR(5000)
     1,B(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)

C---- INPUT WORKING VARIABLES

      READ *,  FS
      READ *,EFC1, ETC, ERATIO
      READ *,DFC1,DTC,DRATIO
      CALL VWXOPT(EVMX,EVMN,FS,EFC1,ETC,ERATIO)
      CALL VWXOPT(DVMX,DVMN,FS,DFC1,DTC,DRATIO)
      READ *,BETA, GAMMA
      PEAK1 = A(-20.)
      PRINT *, ' SNR TEST AT 800 HZ AND SAMPLE RATE = ',FS
      PRINT *, ' EFC1 = ',EFC1,',  ETC = ',ETC,',   ERATIO = ',ERATIO
      PRINT *, ' DFC1 = ',DFC1,',  DTC = ',DTC,',   DRATIO = ',DRATIO
      PRINT *, ' FILTER PARAMETERS ARE, BETA = ',BETA,',  GAMMA = ',GAMM
     1A

C---- GENERATE OUTPUT FILTER COEFFICIENTS

      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- GENERATE INPUT TIME SERIES SAMPLES

      CALL SIGNAL(TSIN,5000,FS,800.,0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME SERIES THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,EFC1,FC2,FC3,ETC,EVMX,EVMN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,DFC1,FC2,FC3,DTC,DVMX,DVMN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE

C---- ADJUST OUTPUT AMPLITUDE SO INPUT POWER = OUTPUT POWER

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN = SQRT (PIN/POUT)
      DO 40 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN
40    CONTINUE

C---- CALCULATE THE NOISE POWER

      DO 50 I = 1,4096
      ERR(I) = TSOUT(I) - TSIN(I)
50    CONTINUE
      CALL POWER(ERR,4096,FS,ERRP)

C---- CALCULATE THE S/N

      SNR = 10. * ALOG10 (PIN / ERRP)

C---- PRINT THE RESULTS

      PRINT *,' THE SIGNAL TO NOISE RATIO = ',SNR
300   CONTINUE
      END
```

APPENDIX S

Signal-to-Noise Ratio vs. Input Signal Frequency Program

```
      PROGRAM SNRRAN(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C----------------------FILTERRED SNR PROGRAM-----------------------
C
C     THIS PROGRAM MEASURES THE SIGNAL-TO-NOISE PERFORMANCE OF A CVSD
C     ENCODER AND DECODER CONNECTED BACK-TO-BACK.  A SINGLE FREQUENCY
C     SINE WAVE IS INPUT TO THE SYSTEM AND THE DIFFERENCE BETWEEN THE
C     OUTPUT AND INPUT COMPUTED.
C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3600 HZ.
C
C     THE PROGRAM IS REPEATED THREE TIMES AS THE VALUE OF THE STEP SIZE
C     RATIO IS CHANGED FROM 32 TO 36 DB IN STEPS OF 2 DB.
C
C
C*********************** VARIABLES *********************************
C
C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3500 HZ.
C
C     SNR = AN ARRAY CONTAINING THE MEASURED SNR VALUES.
C
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES.
C
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME FUNCTION
C        SAMPLES, THE THE FILTER OUTPUT TIME SAMPLES.
C
C     ERR = AN ARRAY CONTAINING THE DIFFERENCE BETWEEN THE OUTPUT SAM-
C        PLES AND THE INPUT SAMPLES.
C
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C
C     FS = THE SAMPLE RATE.
C
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C
C     BETA = THE NORMALIZED 3 DB FREQUENCY OF THE OUTPUT FILTER.  THE
C        FREQUENCY IS NORMALIZED TO THE SAMPLE RATE.
C
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C
C     ICH = THE NUMBER OF TEST FREQUENCIES.
C
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C
C*******************************************************************
C******************** SUBROUTINES USED *****************************
C
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C
C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.
C
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
C
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.
C
C     DECODE1 = THE CVSD DECODING SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.
C
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME FUNTION SAM-
C        PLES USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME FUNC-
C        TION WITH IMPEDENCE = 600 OHMS.
```

```
C-------------------PROGRAM START--------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION FREQ1(100), SNR(100), TSIN(5000), TSOUT(5000), ERR(5000)
     1,B(200), GAIN(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1
C---- INPUT AND PRINT WORKING VARIABLES
      READ *, AMP1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PEAK1 = A(AMP1)
      PRINT *, ' SNR TEST AT ',AMP1,' DBM0 AND ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC
      PRINT *, ' BETA = ',BETA,',  GAMMA = ',GAMMA
C---- GENERATE FILTER COEFFICIENTS
      CALL FLTRGEN(BETA,GAMMA,NP,B)
C---- INITIALZE PLOTTER
      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)
C---- START OF SIGNAL-TO-NOISE LOOP
      DO 1000 NTIMES = 2,8,2
      KN = 0
      ICHAR = ICHAR + 1
      RATIO = 30. + NTIMES
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      DO 300 K = 300,3600,100
      KN = KN + 1
C---- GENERATE TEST SIGNAL FREQUENCY
      FREQ1(KN) = K
C---- GENERATE INPUT TIME FUNCTION SAMPLES
      CALL SIGNAL(TSIN,5000,FS,FREQ1(KN),0.,PEAK1,0.)
C---- PROCESS THE INPUT TIME FUNCTION THROUGH THE CVSD SYSTEM
      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
C---- FILTER THE OUTPUT OF THE DECODER
      CALL FILTER(TSOUT,5000,NP,B)
C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.
      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE
C---- ADJUST OUTPUT AMPLITUDE SO INPUT POWER = OUTPUT POWER
      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN(KN) = SQRT (PIN/POUT)
      DO 40 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN(KN)
40    CONTINUE
C---- CALCULATE THE NOISE POWER
      DO 50 I = 1,4096
      ERR(I) = TSOUT(I) - TSIN(I)
50    CONTINUE
      CALL POWER(ERR,4096,FS,ERRP)
C---- CALCULATE THE S/N
      SNR(KN) = 10. * ALOG10 (PIN / ERRP)
```

```
C---- PLOT THE RESULTS

300   CONTINUE
      IF (ICHAR .GT. 0) GO TO 900
      CALL PLOT(12.,8.,-3)
      CALL LGSCAL(FREQ1,10.,KN)
      CALL SCALE(SNR,6.,KN,1)
      CALL LGAXIS(0.,0.,14HFREQUENCY (HZ),-14,10.,0.,FREQ1(KN+1),FREQ1(K
     1N+2))
      CALL AXIS(0.,0.,8HSNR (DB),8,6.,90.,SNR(KN+1),SNR(KN+2))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LGLINE(FREQ1,SNR,KN,10,ICHAR,-1)
1000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX T

Mismatched Signal-to-Noise Ratio vs. Input Signal Frequency

```
      PROGRAM MMSNRF(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)

C-------- -----------MISMATCHED SNR PROGRAM---------------------------

C     THIS PROGRAM MEASURES THE SIGNAL-TO-NOISE PERFORMANCE OF A CVSD
C     ENCODER AND DECODER CONNECTED BACK-TO-BACK.  A SINGLE FREQUENCY
C     SINE WAVE IS INPUT TO THE SYSTEM AND THE DIFFERENCE BETWEEN THE
C     OUTPUT AND INPUT COMPUTED.

C     THE CALCULATIONS ARE PERFORMED AS THE VALUE OF THE ENCODER STEP
C     SIZE RATIO IS KEPT CONSTANT AND THE DECODER STEP SIZE RATIO IS
C     CHANGED.  THE ENCODER RATIO IS 32 DB, WHILE THE DECODER IS STEPPED
C     FROM 32 TO 36 DB IN STEPS OF 2 DB.

C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3000 HZ.
C
C************************ VARIABLES **********************************

C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3300 HZ.

C     SNR = AN ARRAY CONTAINING THE MEASURED SNR VALUES.

C     TSIN = AN ARRAY CONTAINING THE INPUT TIME SERIES SAMPLES.

C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME SERIES
C        SAMPLES, THEN THE OUTPUT TIME SERIES SAMPLES OF THE FIR FILTER.

C     ERR = AN ARRAY CONTAINING THE DIFFERENCE BETWEEN THE OUTPUT SAM-
C        PES AND THE INPUT SAMPLES.

C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.

C     TIME = AN ARRAY CONTAINING THE TIME THAT THE FIRST 200 SAMPLES
C        ARE TAKEN SO THAT THEY MAY BE PLOTTED.

C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER

C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.

C     FS = THE SAMPLE RATE.

C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.

C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.

C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.

C     BETA = THE NORMALIZED 3 DB FREQUENCY OF THE OUTPUT FILTER.  THE
C        FREQUENCY IS NORMALIZED TO THE SAMPLE RATE.

C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.

C     NP = THE NUMBER OF FILTER COEFFICIENTS.

C     KN = THE NUMBER OF TEST FREQUENCIES.

C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.

C*********************************************************************
C******************** SUBROUTINES USED *******************************

C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.

C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.

C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.

C     ENCODE1 = THE CVSD ENCODER SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.

C     DECODE1 = THE CVSD DECODING SUBROUTINE WITH A SINGLE ROLL-OFF
C        FREQUENCY IN THE PRIMARY INTEGRATOR.
```

```
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME SERIES SAMPLES
C        USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.

C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME SERIES
C        WITH IMPEDENCE = 600 OHMS.

C     PLTIME = A LINEAR PLOTTING ROUTINE TO PLOT SIGNAL AMPLITUDE VS.
C        TIME.

C*************************************************************************
C*************************************************************************

C-------------------PROGRAM START------------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION FREQ1(100), SNR(100), TSIN(5000), TSOUT(5000), ERR(5000)
     1,B(200), TIME(200), GAIN(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1

C---- INPUT WORKING VARIABLES

      READ *, AMP1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PEAK1 = A(AMP1)
      PRINT *, ' SNR TEST AT ',AMP1,' DBM0 AND ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC,' AND RATIO = ',RATIO
      PRINT *, ' BETA = ',BETA,',  GAMMA = ',GAMMA

C---- GENERATE OUTPUT FILTER COEFFICIENTS

      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- INITIALZE PLOTTER

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)

C---- START OF SIGNAL-TO-NOISE LOOP

      DO 1000 NTIMES = 2,6,2
      KN = 0
      ICHAR = ICHAR + 1
      RATIO = 30. + NTIMES
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      IF (ICHAR .GT. 0) GO TO 3
      EVMX = VMAX
      EVMN = VMIN
3     CONTINUE

      DO 300 K = 300,3600,100
      KN = KN + 1

C---- GENERATE TEST SIGNAL FREQUENCY

      FREQ1(KN) = K

C---- GENERATE INPUT TIME SERIES SAMPLES

      CALL SIGNAL(TSIN,5000,FS,FREQ1(KN),0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME SERIES THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,EVMX,EVMN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE

C---- ADJUST OUTPUT AMPLITUDE SO INPUT POWER = OUTPUT POWER

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN(KN) = SQRT (PIN/POUT)
      DO 40 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN(KN)
40    CONTINUE
```

```
C---- CALCULATE THE NOISE POWER

      DO 50 I = 1,4096
      ERR(I) = TSOUT(I) - TSIN(I)
50    CONTINUE
      CALL POWER(ERR,4096,FS,ERRP)

C---- CALCULATE THE S/N

      SNR(KN) = 10. * ALOG10 (PIN / ERRP)

C---- PRINT AND PLOT THE RESULTS

      WRITE(6,600) FREQ1(KN), SNR(KN), GAIN(KN)
600   FORMAT(20X,F10.1,5X,F10.6,5X,F10.6)
300   CONTINUE
      IF (ICHAR .GT. 0) GO TO 900
      CALL LGSCAL(FREQ1,10.,KN)
      CALL SCALE(SNR,6.,KN,1)
      CALL LGAXIS(0.,0.,14HFREQUENCY (HZ),-14,10.,0.,FREQ1(KN+1),FREQ1(K
     1N+2))
      CALL AXIS(0.,0.,8HSNR (DB),8,6.,90.,SNR(KN+1),SNR(KN+2))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LGLINE(FREQ1,SNR,KN,10,ICHAR,-1)
1000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX U

Signal-to-Noise Ratio vs. Input Signal Power Program

```
      PROGRAM SNRD(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C----------------------SNR VS. INPUT POWER PROGRAM----------------------
C     THIS PROGRAM MEASURES THE SIGNAL-TO-NOISE PERFORMANCE OF A CVSD
C     ENCODER AND DECODER CONNECTED BACK-TO-BACK.  A SINGLE FREQUENCY
C     SINE WAVE IS INPUT TO THE SYSTEM AND THE DIFFERENCE BETWEEN THE
C     OUTPUT AND INPUT COMPUTED.
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3200 HZ.
C     THE INPUT POWER IS VARIED FROM -40 DBM0 TO 0 DBM0.  THE PROGRAM
C     IS REPEATED AS THE STEP SIZE RATIO IS VARIED FROM 32 DB TO 36 DB.
C     THE RESULTS ARE THEN PLOTTED ON A SINGLE GRAPH.
C*************************** VARIABLES **********************************
C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3000 HZ.
C     SNR = AN ARRAY CONTAINING THE MEASURED SNR VALUES.
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME SERIES SAMPLES.
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME SERIES
C        SAMPLES, THEN THE OUTPUT TIME SERIES SAMPLES OF THE FIR FILTER.
C     ERR = AN ARRAY CONTAINING THE DIFFERENCE BETWEEN THE OUTPUT SAM-
C        PLES AND THE INPUT SAMPLES.
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C     TIME = AN ARRAY CONTAINING THE TIME THAT THE FIRST 200 SAMPLES
C        ARE TAKEN SO THAT THEY MAY BE PLOTTED.
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE.
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C     BETA = THE MIDPOINT OF THE OUTPUT FILTER TRANSITION BAND.
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C************************************************************************
C********************* SUBROUTINES USED *********************************
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE
C     DECODE1 = THE CVSD DECODER SUBROUTINE
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME SERIES SAMPLES
C        USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME SERIES
C        WITH IMPEDENCE = 600 OHMS.
C************************************************************************
```

```
C*********************************************************************
C-----------------PROGRAM START---------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION AMP1(200), SNR(200), TSIN(5000), TSOUT(5000), ERR(5000)
     1,B(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1

C---- INPUT AND PRINT THE WORKING VARIABLES

      READ *, FREQ1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PRINT *, ' SNR TEST AT ',FREQ1,' DBM0 AND ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC,' AND RATIO = ',RATIO
      PRINT *,' FILTER PARAMETERS ARE, BETA = ',BETA,',GAMMA = ',GAMMA

C---- GENERATE OUTPUT FILTER COEFFICIENTS

      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- INITIALZE PLOTTER

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)

C---- START LOOP

      DO 1000 NR = 0,6,2
      RATIO = 33. + NR
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)

C---- START OF SIGNAL-TO-NOISE LOOP

      DO 300 K = 1,100
      AMP1(K) = -40. + .4 * K
      PEAK1 = A(AMP1(K))


C---- GENERATE INPUT TIME SERIES SAMPLES

      CALL SIGNAL(TSIN,5000,FS,FREQ1,0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME SERIES THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
 30   CONTINUE

C---- ADJUST THE OUTPUT SIGNAL AMPLITUDE SO, OUTPUT POWER = INPUT POWER

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN = SQRT (PIN/POUT)
      DO 40 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN
 40   CONTINUE

C---- CALCULATE THE NOISE POWER

      DO 50 I = 1,4096
      ERR(I) = TSOUT(I) - TSIN(I)
 50   CONTINUE
      CALL POWER(ERR,4096,FS,ERRP)

C---- CALCULATE THE S/N

      SNR(K) = 10. * ALOG10 (PIN / ERRP)
      IF (ICHAR .LT. 6) GO TO 200
      IF (SNR(K) .LT. SNR(101)) SNR(K) = SNR(101)
 200  CONTINUE
 300  CONTINUE
```

```
C---- PRINT AND PLOT THE RESULTS

      ICHAR = ICHAR + 1
      IF (ICHAR .GT. 0) GO TO 900
      CALL SCALE(AMP1,10.,100,1)
      CALL SCALE(SNR,6.,100,1)
      CALL AXIS(0.,0.,16HAMPLITUDE (DBM0),-16,10.,0.,AMP1(101),AMP1(102)
     1)
      CALL AXIS(0.,0.,8HSNR (DB),8,6.,90.,SNR(101),SNR(102))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LINE(AMP1,SNR,100,1,10,ICHAR)
1000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX V

Mismatched Signal-to-Noise Ratio vs. Input Signal Power Program

```
      PROGRAM MMSNRD(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C-----------------MISMATCHED SNR VS. INPUT POWER PROGRAM---------------
C
C     THIS PROGRAM MEASURES THE SIGNAL-TO-NOISE PERFORMANCE OF A CVSD
C     ENCODER AND DECODER CONNECTED BACK-TO-BACK.  A SINGLE FREQUENCY
C     SINE WAVE IS INPUT TO THE SYSTEM AND THE DIFFERENCE BETWEEN THE
C     OUTPUT AND INPUT COMPUTED.
C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3600 HZ.
C
C     THE INPUT POWER IS VARIED FROM -40 DBM0 TO 0 DBM0.  THE PROGRAM
C     IS REPEATED AS THE STEP SIZE RATIO IS VARIED FROM 22 DB TO 25 DB
C     IN THE DECODER, WHILE THE ENCODER STEP SIZE RATIO IS HELD CON-
C     STANT.
C
C     THE RESULTS ARE THEN PLOTTED ON A SINGLE GRAPH.
C
C
C************************* VARIABLES ***********************************
C
C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3000 HZ.
C
C     SNR = AN ARRAY CONTAINING THE MEASURED SNR VALUES.
C
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME SERIES SAMPLES.
C
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME SERIES
C        SAMPLES, THEN THE OUTPUT TIME SERIES SAMPLES OF THE FIR FILTER.
C
C     ERR = AN ARRAY CONTAINING THE DIFFERENCE BETWEEN THE OUTPUT SAM-
C        PLES AND THE INPUT SAMPLES.
C
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C
C     TIME = AN ARRAY CONTAINING THE TIME THAT THE FIRST 200 SAMPLES
C        ARE TAKEN SO THAT THEY MAY BE PLOTTED.
C
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C
C     FS = THE SAMPLE RATE.
C
C**   NOTE: A 'D' OR 'E' PREFIX ON THE FOLLOWING VARIABLES INDICATES
C        THAT THE VARIABLE IS USED BY THE DECODER OR ENCODER,RESPECTIVE-
C        LY. VMAX AND VMIN MAY BE CONTRACTED TO VMX AND VMN, RESPECTIVE-
C        LY.
C
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C
C     BETA = THE NORMALIZED MIDPOINT OF THE TRANSITION BAND OF THE OUT-
C        PUT LOW PASS FILTER.
C
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C
C     KN = THE NUMBER OF TEST FREQUENCIES.
C
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C
C***********************************************************************
C******************** SUBROUTINES USED *********************************
C
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C
C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.
C
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
```

```
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE

C     DECODE1 = THE CVSD DECODER SUBROUTINE

C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME SERIES SAMPLES
C        USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.

C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME SERIES
C        WITH IMPEDENCE = 600 OHMS.

C******************************************************************************
C******************************************************************************

C-------------------PROGRAM START-----------------------------------------

C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION AMP1(200), SNR(200), TSIN(5000), TSOUT(5000), ERR(5000)
     1,B(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1

C---- INPUT AND PRINT WORKING VARIABLES

      READ *, FREQ1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PRINT *, ' SNR TEST AT ',FREQ1,' DBM0 AND ',FS,' SPS'
      PRINT *,' WITH TC = ',TC
      PRINT *,' FILTER PARAMETERS ARE, BETA = ',BETA,',GAMMA = ',GAMMA

C---- GENERATE OUTPUT FILTER COEFFICIENTS

      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- INITIALZE PLOTTER

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)

C---- START LOOP

      DO 1000 NR = 2,6,2
      RATIO = 30. + NR
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      IF ( ICHAR .GE. 0) GO TO 6
      EVMX = VMAX
      EVMN = VMIN
6     CONTINUE

C---- START OF SIGNAL-TO-NOISE LOOP

      DO 300 K = 1,100
      AMP1(K) = -40. + .4 * K
      PEAK1 = A(AMP1(K))


C---- GENERATE INPUT TIME SERIES SAMPLES

      CALL SIGNAL(TSIN,5000,FS,FREQ1,0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME SERIES THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,EVMX,EVMN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE

C---- ADJUST OUTPUT SIGNAL AMPLITUDE SO, OUTPUT POWER = INPUT POWER

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN = SQRT (PIN/POUT)
      DO 40 I = 1,4096
      TSOUT(I) = TSOUT(I) * GAIN
40    CONTINUE
```

```
C---- CALCULATE THE NOISE POWER

      DO 50 I = 1,4098
      ERR(I) = TSOUT(I) - TSIN(I)
50    CONTINUE
      CALL POWER(ERR,4096,FS,ERRP)

C---- CALCULATE THE S/N

      SNR(K) = 10. * ALOG10 (PIN / ERRP)
      IF (ICHAR .LT. 0) GO TO 60
      IF (SNR(K) .LT. SNR(101)) SNR(K) = SNR(101)
60    CONTINUE
300   CONTINUE

C---- PRINT AND PLOT THE RESULTS

      ICHAR = ICHAR + 1
      IF (ICHAR .GT. 0) GO TO 900
      CALL SCALE(AMP1,10.,100,1)
      CALL SCALE(SNR,6.,100,1)
      CALL AXIS(0.,0.,10HAMPLITUDE (DBM0),-16,10.,0.,AMP1(101),AMP1(102)
     1)
      CALL AXIS(0.,0.,6HSNR (DB),8,6.,90.,SNR(101),SNR(102))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LINE(AMP1,SNR,100,1,10,ICHAR)
1000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX W

Flat Weighted Frequency Response Program

```
      PROGRAM DIFGAIN(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C------------------DIFFERNTIAL GAIN PROGRAM----------------------------
C
C     THIS PROGRAM MEASURES THE SYSTEM GAIN VS. FREQUENCY RESPONSE FOR
C     A CVSD DIGITAL/ANALOG SYSTEM CONNECTED BACK-TO-BACK.
C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3600 HZ.
C
C************************ VARIABLES ***********************************
C
C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3300 HZ.
C
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES.
C
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME FUNCTION
C        SAMPLES, THEN THE OUTPUT TIME FUNCTION SAMPLES OF THE FIR FIL-
C        TER.
C
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C
C     FS = THE SAMPLE RATE.
C
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C
C     BETA = THE NORMALIZED CENTER FREQUENCY OF THE OUTPUT FILTER TRANS-
C        ITION BAND.
C
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C
C     KN = THE NUMBER OF TEST FREQUENCIES.
C
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C
C*********************************************************************
C********************* SUBROUTINES USED ******************************
C
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C
C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.
C
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
C
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE
C
C     DECODE1 = THE CVSD DECODER SUBROUTINE
C
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME FUNCTION SAM-
C        PLES USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME FUNC-
C        TION WITH IMPEDENCE = 600 OHMS.
C
C*********************************************************************
C*********************************************************************
```

```
C------------------PROGRAM START-----------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS

      DIMENSION FREQ1(100),  TSIN(5000), TSOUT(5000)
     1,B(200),  GAIN(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1

C---- INPUT AND PRINT THE WORKING VARIABLES

      READ *, AMP1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PRINT *, ' SNR TEST AT ',AMP1,' DBM0 AND ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC
      PRINT *,' OUTPUT FILTER PARAMETERS ARE, BETA = ',BETA,' GAMMA ',GA
     1MMA

C---- GENERATE OUTPUT FILTER COEFFICIENTS

      CALL FLTRGEN(BETA,GAMMA,NP,B)

C---- INITIALZE PLOTTER

      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)

C---- START LOOP

      DO 2000 NTIMES = 2,6,2
      ICHAR = ICHAR + 1
      RATIO = 30. + NTIMES
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      KN = 0

C---- START OF SIGNAL-TO-NOISE LOOP

      DO 300 K = 300,3600,100
      KN = KN + 1

C---- GENERATE TEST SIGNAL

      FREQ1(KN) = K
      PEAK1 = A(AMP1)
      CALL SIGNAL(TSIN,5000,FS,FREQ1(KN),0.,PEAK1,0.)

C---- PROCESS THE INPUT TIME FUNCTION THROUGH THE CVSD SYSTEM

      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)

C---- FILTER THE OUTPUT OF THE DECODER

      CALL FILTER(TSOUT,5000,NP,B)

C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.

      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE

C---- FIND SYSTEM GAIN

      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN(KN) = 10. * ALOG10(POUT/PIN)
300   CONTINUE
```

```
C---- ADJUST GAIN VALUES TO 800 HZ REFERERENCE

      DO 6 I = 1,KN
      IF (FREQ1(I) .EQ. 800.) REFGAIN = GAIN(I)
6     CONTINUE
      DO 8 I = 1,KN
      GAIN(I) = GAIN(I) - REFGAIN
8     CONTINUE

C---- PLOT THE RESULTS

      IF (ICHAR .GT. 0) GO TO 900
      CALL SCALE(GAIN,6.,KN,1)
      CALL LGSCAL(FREQ1,10.,KN)
      CALL LGAXIS(0.,0.,14HFREQUENCY (HZ),-14,10.,0.,FREQ1(KN+1),FREQ1(K
     1N+2))
      CALL AXIS(0.,0.,22HDIFFERENTIAL GAIN (DB),22,6.,90.,GAIN(KN+1),
     1GAIN(KN+2))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LGLINE(FREQ1,GAIN,KN,10,ICHAR,-1)
2000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX X

Mismatched Flat Weighted Frequency Response Program

```
      PROGRAM MMDGAIN(INPUT,OUTPUT,TAPE6=OUTPUT,PLOT)
C---------------MISMATCHED SYSTEM GAIN RESPONSE-----------------------
C     THIS PROGRAM MEASURES THE SYSTEM GAIN VS. FREQUENCY RESPONSE FOR
C     A CVSD DIGITAL/ANALOG SYSTEM CONNECTED BACK-TO-BACK.
C
C     A MAXIMALLY FLAT LINEAR PHASE FIR FILTER IS PLACED ON THE OUTPUT
C     OF THE DECODER TO REMOVE SIGNAL COMPONENTS ABOVE 3600 HZ.
C
C************************ VARIABLES **********************************
C     FREQ1 = AN ARRAY CONTAINING THE FREQUENCIES THAT THE SNR HAS BEEN
C        MEASURED AT.  THE RANGE IS 300 HZ TO 3300 HZ.
C     TSIN = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES.
C     TSOUT = AN ARRAY CONTAINING FIRST THE DECODER OUTPUT TIME FUNCTION
C        SAMPLES, THEN THE OUTPUT TIME FUNCTION SAMPLES OF THE FIR FIL-
C        TER.
C     B = AN ARRAY CONTAINING THE FILTER COEFFICIENTS.
C     BINOUT = AN ARRAY CONTAINING THE BINARY OUTPUT OF THE CVSD ENCODER
C     AMP1 = THE AMPLITUDE OF THE TEST SIGNAL IN DBM0.
C     FS = THE SAMPLE RATE.
C     FC1, FC2, FC3 = THE ROLL-OFF FREQUENCIES OF THE PRIMARY INTEGRA-
C        TORS.
C     TC = THE TIME CONSTANT OF THE SYLLABIC FILTERS.
C     VMAX & VMIN = THE MAXIMUM AND MINIMUM INPUTS TO THE SYLLABIC FIL-
C        TER.
C     BETA = THE NORMALIZED CENTER FREQUENCY OF THE OUTPUT FILTER TRANS-
C        ITION BAND.
C     GAMMA = THE NORMALIZED WIDTH OF THE ROLL-OFF REGION OF THE OUTPUT
C        FILTER.  THE REGION IS THE FREQUENCY BAND BETWEEN THE 95% AND
C        5% OUTPUT AMPLITUDES.
C     PEAK1 = THE MAXIMUM AMPLITUDE OF THE TEST SIGNAL IN VOLTS.
C     NP = THE NUMBER OF FILTER COEFFICIENTS.
C     NI = THE NUMBER OF TEST FREQUENCIES.
C     DC = THE DUTY CYCLE OF THE SLOPE OVERLOAD DETECTOR.
C*********************************************************************
C******************** SUBROUTINES USED *******************************
C     FLTRGEN = THE SUBROUTINE THAT GENERATES THE OUTPUT FILTER COEFFI-
C        CIENTS.
C     PLOT, SCALE, AXIS, RECT, LINE, PLOTE, = CALCOMP PLOTTING ROUTINES.
C     SIGNAL = THE TEST SIGNAL GENERATOR.  PRODUCES SAMPLES OF SINUSOI-
C        DAL WAVES WITH AT MOST TWO FREQUENCY COMPONENTS.
C     ENCODE1 = THE CVSD ENCODER SUBROUTINE
C     DECODE1 = THE CVSD DECODER SUBROUTINE
C     FILTER = THE SUBROUTINE THAT FILTERS THE INPUT TIME FUNCTION SAM-
C        PLES USING THE FILTER COEFFICIENTS GENERATED BY FLTRGEN.
C     POWER = A ROUTINE TO CALCULATE THE POWER IN A SAMPLED TIME FUNC-
C        TION WITH IMPEDENCE = 600 OHMS.
C*********************************************************************
C*********************************************************************
```

```
C-------------------PROGRAM START---------------------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION FREQ1(100),  TSIN(5000), TSOUT(5000)
     1,B(200),  GAIN(200)
      INTEGER BINOUT(5000)
      A(DBM0) = SQRT(10. **((DBM0 -4.)/10.) * .001 * 600.) * SQRT(2.)
      ICHAR = -1
C---- INPUT AND PRINT THE WORKING VARIABLES
      READ *, AMP1, FS
      READ *,FC1, TC
      READ *,BETA, GAMMA
      PRINT *, ' SNR TEST AT ',AMP1,' DBM0 AND ',FS,' BPS'
      PRINT *, ' WITH TC = ',TC
      PRINT *,' OUTPUT FILTER PARAMETERS ARE, BETA = ',BETA,' GAMMA ',GA
     1MMA
C---- GENERATE OUTPUT FILTER COEFFICIENTS
      CALL FLTRGEN(BETA,GAMMA,NP,B)
C---- INITIALZE PLOTTER
      CALL FACTOR(.5)
      CALL PLOT(2., 2., -3)
C---- START LOOP
      DO 2000 NTIMES = 2,6,2
      ICHAR = ICHAR + 1
      RATIO = 30. + NTIMES
      CALL VMAXOPT(VMAX,VMIN,FS,FC1,TC,RATIO)
      KN = 0
      IF (ICHAR .GT. 0) GO TO 100
      EVMX = VMAX
      EVMN = VMIN
100   CONTINUE
C---- START OF SIGNAL-TO-NOISE LOOP
      DO 300 K = 300,3600,100
      KN = KN + 1
C---- GENERATE TEST SIGNAL
      FREQ1(KN) = K
      PEAK1 = A(AMP1)
      CALL SIGNAL(TSIN,5000,FS,FREQ1(KN),0.,PEAK1,0.)
C---- PROCESS THE INPUT TIME FUNCTION THROUGH THE CVSD SYSTEM
      CALL ENCODE1(TSIN,BINOUT,5000,FS,FC1,FC2,FC3,TC,EVMX,EVMN,DC)
      CALL DECODE1(BINOUT,TSOUT,5000,FS,FC1,FC2,FC3,TC,VMAX,VMIN,DC)
C---- FILTER THE OUTPUT OF THE DECODER
      CALL FILTER(TSOUT,5000,NP,B)
C---- DELAY THE INPUT SIGNAL START TO CORRESPOND TO THE
C     FILTERRED OUTPUT.
      DO 30 ID = 1,4096
      KD = 200 + ID
      TSIN(ID) = TSIN(KD)
30    CONTINUE
C---- FIND SYSTEM GAIN
      CALL POWER(TSIN,4096,FS,PIN)
      CALL POWER(TSOUT,4096,FS,POUT)
      GAIN(KN) = 10. * ALOG10(POUT/PIN)
300   CONTINUE
C---- ADJUST GAIN VALUES TO 800 HZ REFERERENCE
```

```
      DO 6 I = 1,KN
      IF (FREQ1(I) .EQ. 800.) REFGAIN = GAIN(I)
6     CONTINUE
      DO 8 I = 1,KN
      GAIN(I) = GAIN(I) - REFGAIN
8     CONTINUE
C---- PLOT THE RESULTS
      IF (ICHAR .GT. 0) GO TO 900
      CALL SCALE(GAIN,6.,KN,1)
      CALL LGSCAL(FREQ1,10.,KN)
      CALL LGAXIS(0.,0.,14HFREQUENCY (HZ),-14,10.,0.,FREQ1(KN+1),FREQ1(K
     1N+2))
      CALL AXIS(0.,0.,22HDIFFERENTIAL GAIN (DB),22,6.,90.,GAIN(KN+1),
     1GAIN(KN+2))
      CALL RECT(0.,0.,6.,10.,0.,3)
900   CONTINUE
      CALL LGLINE(FREQ1,GAIN,KN,10,ICHAR,-1)
2000  CONTINUE
      CALL PLOTE(N)
      END
```

APPENDIX Y

Test Signal Generator

```
      SUBROUTINE SIGNAL(OUTPUT,N,FS,FREQ1,FREQ2,AMP1,AMP2)
C**********************************************************************
C     THIS SUBROUTINE GENERATES A  TEST SIGNAL COMPOSED OF UP
C     TO TWO SINE WAVES OF DIFFERENT FREQUENCIES AND AMPLITUDES.
C*********************** VARIABLES ************************************
C     OUTPUT = AN ARRAY CONTAINING THE OUTPUT TIME FUNCTION SAMPLES
C     N = THE NUMBER OF SAMPLES OF THE TIME FUNCTION DESIRED
C     FS = THE SAMPLE RATE IN KB/S
C     FREQ1 = THE FREQUENCY OF THE FIRST SIGNAL COMPONENT
C     FREQ2 = THE FREQUENCY OF THE SECOND SIGNAL COMPONENT
C     AMP1 = THE PEAK AMPLITUDE OF THE FIRST SIGNAL COMPONENT
C     AMP2 = THE PEAK AMPLITUDE OF THE SECOND SIGNAL COMPONENT
C**********************************************************************
C**********************************************************************
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION OUTPUT(N)
      DATA PI/3.1415926538/
C---- GENERATE OUTPUT SAMPLES OF TEST SIGNAL
      DO 50 I = 1,N
      OUTPUT(I) = AMP1 * SIN( 2. * PI * FREQ1 / FS * I)
     1+ AMP2 * SIN (2. * PI * FREQ2 / FS * I)
50    CONTINUE
      RETURN
      END
```

APPENDIX Z

Signal Power Calculation Subroutine

```
      SUBROUTINE POWER(X,N,FS,P)
C-----------------------AVERAGE POWER SUBROUTINE-----------------------
C     THIS SUBROUTINE CALCULATES THE AVERAGE POWER OF THE INPUT TIME
C     FUNCTION SAMPLES ACROSS A 600 OHM IMPEDENCE.  THE INPUT TIME FUNC-
C     TION SAMPLES ARE INPUT TO THE SUBROUTINE AS A 1 X N ARRAY.
C*********************** VARIABLES **********************************
C     X = AN ARRAY CONTAINING THE INPUT TIME FUNCTION SAMPLES
C     N = THE NUMBER OF SAMPLES TO BE PROCESSED
C     FS = THE SAMPLE RATE
C     P = THE CALCULATED SIGNAL POWER
C*********************************************************************
C-------------------------SUBROUTINE START-------------------------
C---- INITIALIZE VARIABLES AND ARRAYS
      DIMENSION X(N)
      SUM = 0.
C---- SUM THE SQUARES OF THE INPUT TIME FUNCTION SAMPLES
      DO 10 I = 1,N
      SUM = SUM + X(I) ** 2.
10    CONTINUE
C---- CALCULATE THE AVERAGE POWER ACROSS A 600 OHM IMPEDENCE
      P = SUM / N / 600.
      RETURN
      END
```

Vita

Jeffrey Allan Iersch was born 20 January 1948 in Milwaukee, Wisconsin. After graduating from New Berlin High School in 1966, he began undergraduate study in electrical engineering at Michigan Technological University. Captain Iersch received his Bachelor of Science degree in electrical engineering and was commissioned in the Air Force in June 1971. His entire Air Force career has been as an engineer working with the military long-haul communications system, first at Headquarters Air Force Communications Service, then in Athens, Greece with the 2140th Communications Group, and finally at the Headquarters Northern Communications Area at Griffiss AFB, New York. In June 1979 Captain Iersch began study at the Air Force Institute of Technology for a Master of Science degree in electrical engineering.

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A computer model of the continuously variable slope delta voice encoding system specified in the draft STANAG on "Analogue/Digital Conversion of Speech Signals for Tactical, Digital, Area Communication Systems", dated June 1978, is developed and implemented in FORTRAN IV. The model's performance is then characterized in terms of idle channel noise, total harmonic distortion, intermodulation distortion, signal-to-noise ratio, and frequency response. For each of these attributes, the system's performance is presented graphically and compared to the criteria established

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in the draft standard. The model is then exercised by varying the system parameters to the limits imposed by the standard and the resulting performance compared to the previously determined ideal system performance. The results show that the performance characteristics measured are most sensitive to the primary integrator response and output filter response when the system parameters are restricted to the range allowed by the draft NATO standard.

END

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